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Coming Next Month

• We start the new year with a little more news about digital audio. Depending on who you read, it may be the greatest—or worst—thing to happen to recording since the invention of the microphone. Next month, we'll have a look at the latest in the digital domain, courtesy of some authors with contrasting views on the subject.

• The Cumulative Index, normally seen in December issues, will appear instead in the January 1981 issue.

About The Cover

• On this month's cover, we see some creative microphone placement, courtesy of photographer Robert Wolsch.

In the foreground, an Electro-Voice RE-20, Shure SM-59 and SM-78, and a Beyer M-160. The middle row shows a Ramsa WM-8100, an Audio-Technica ATM-11SM and a pair of Shure SM-81s. In the background, a Sony C-48, a Beyer M-500 and an Audio-Technica AT-815.

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db Letters

To The Editor:

I attended the Natural Stereo Recording Techniques workshop at Eau Claire, Wisconsin and found that people were using the M-S recording technique with the M opened up to omni or even figure-8 modes. What are the theoretical mic patterns which result from such matrix applications? I find to my chagrin that I can't figure it out.

Thomas Ammons
Glenshaw, Pa.

db replies:

In M-S recording, the M mic is usually cardioid *and the S is a figure-8. For details about what happens when M and S are combined, see our Application Notes in this issue.*

ON REMOVING DISTORTION

To The Editor:

In Norman Crowhurst's reply (db June 1980) to my letters to the editor of November 1979 and April 1980, he states that I credit myself with the original idea for nonlinear distortion by complementary distortion. I wish to point out that I have never claimed such originality and, in my papers on the subject, I refer to earlier work by others in the field. I was, however, probably the first to publish explicit mathematical treatments of the possibilities and limitations of the method.

Although Crowhurst now agrees that his original statement (which occasioned this exchange of letters), "In fact, however distortion gets in, you cannot take it out again," may be modified that one can in fact *reduce* (nonlinear) distortion

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provided one has sufficient information about the distorting mechanism, he still stated in his June 1980 letter that distortion could not be eliminated by complementary distortion techniques. I have now learned, through direct correspondence with Mr. Crowhurst, that he has not had a chance, since the beginning of this exchange of letters to the editor in November 1979, to look at my papers on complementary distortion, referenced in the November 1979 letter. The papers do, in fact, show how it is theoretically possible to entirely eliminate nonlinear distortion within at least a limited input signal amplitude range. In actual practice, of course, one can only reduce such distortion by an arbitrary amount, and the greater the reduction wanted the greater the circuit complexity required. You can, therefore, take the distortion out to whatever degree needed.

J. ROSS MACDONALD
U. of North Carolina

BARRY BLESSER

The Realities of Digital Technology

• We are now ready to look at a digital audio system in terms of the actual performance which we can expect, using real components. Up to this point, our discussions have mainly been about an ideal system with perfect components. But alas, our perfect digital system actually produces not-very-perfect results, due to defects in the components. To be charitable, we should call them "limitations," since nothing is ever perfect. But because we are examining digitization, rather than analog technology, our inclination is to be less forgiving about defects in the digital domain. Hence, our criterion for "acceptable" is often much higher.

We will begin our discussion about digital audio defects with the DAC (digital-to-audio converter), since this element is used in both encoding and decoding. But, before we begin, we need to resolve a semantic difficulty with the word DAC. The problem is that there is a circuit component which converts a digital word to an analog voltage, and this component is used in both the encoding (analog-to-digital) and decoding (digital-to-analog) parts of the digital audio system. From this point forward, we will refer to the DAC as the component; and A/D and D/A as the encoding and decoding system. Both contain the DAC component.

THE DAC COMPONENT

In previous discussions, we demonstrated the operation of the DAC component by showing the relationship between the digital number (word) inputs and the analog (voltage) outputs. Each word corresponds to a specific voltage, and a uniform DAC has a consistent voltage difference between neighboring digital words. With a 10-bit DAC component having a ±10 volt output range, there will be 1024 quantization levels, each separated by 20 millivolts. We will thus expect a quantization level at −30 mV, −10 mV +10mV, +30 mV, etc. However, in a real DAC component, the quantization levels may not be quite where we expect them to be; there will be an error. For example, the quantization level which was expected to be at +10 mV might actually be at +14 mV. This is a positive error of 4 mV. The quantization level which should have been at +30 mV might actually be at +18 mV, a negative error of −12 mV.

One way of specifying the quality of a converter is to specify the largest error between the DAC component and an idealized perfect DAC having the same number of bits. Since millivolts of error is an inconvenient metric, we typically measure the error as a ratio of the maximum error voltage to the ideal difference

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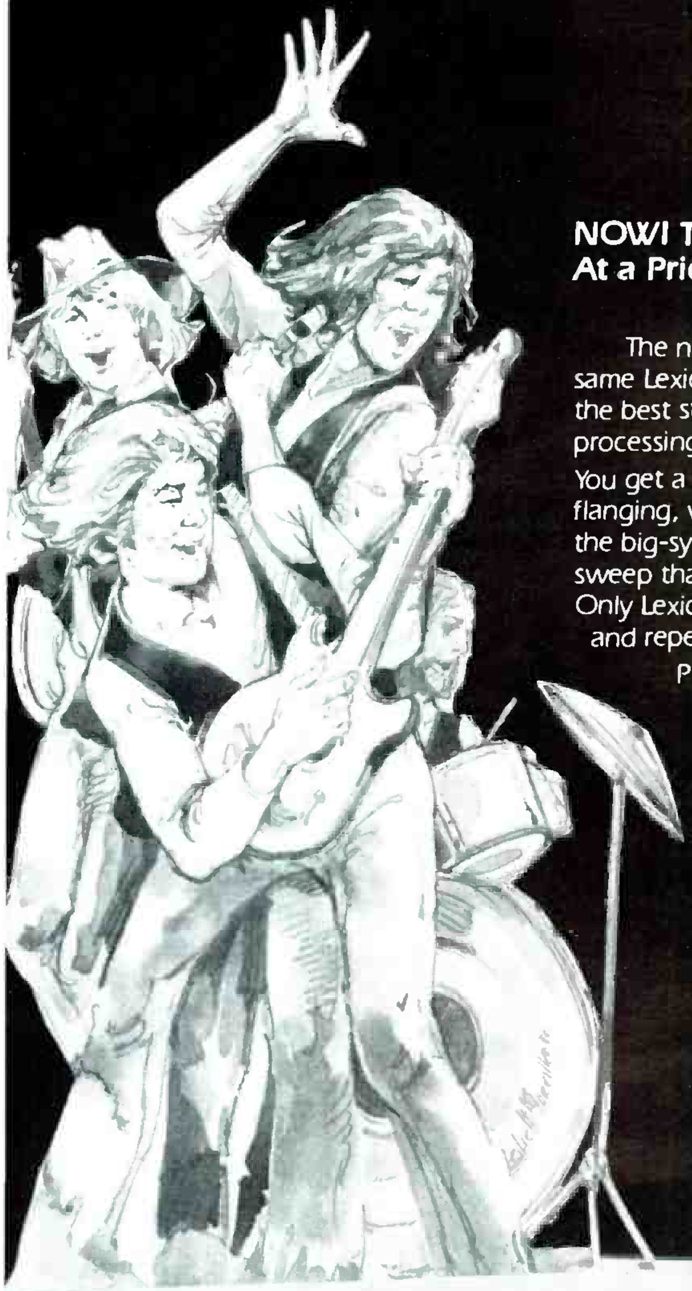

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in quantization levels. In our example, neighboring levels are intended to be 20 mV apart. This is defined as an LSB (least significant bit), since a change in the lowest-level bit will change the output by this amount. If the worst-case quantization level were in error by 4 mV, we would say that the maximum error was 1/5 LSB. If the worst-case error were 30 mV, we would say that the converter was accurate to 1.5 LSBs. In this discussion, accuracy is the same as linearity, since an inaccuracy produces a non-linear relationship between digital input and analog output. The absolute linearity error is the name for this kind of inaccuracy.

Until recently, most manufacturers of DAC components specified a worst-case absolute non-linearity of ±0.5 LSBs. This criterion was selected because it also implied that the DAC was monotonic. Monotonic means that a larger digital input always produces a larger analog output. Let us examine this case in detail. With a ± 0.5 LSB error, a quantization level at + 10 mV could be any place between 0 mV and + 20 mV, and a quantization level at 30 mV could be any place between 20 mV and 40 mV. Notice that these two levels could produce the same output of + 20 mV if the higher one had a maximum negative error and the lower one had a maximum positive error. If the tolerance on the DAC component had been ± 1 LSB, then these two levels could overlap. In other words, the higher one could have been at + 10 mV and the lower one at + 30 mV. Hence, monotonicity, or ± 0.5 LSB, is the same specification. In the A/D encoder, the corresponding name for this criterion is "no missing codes," but this discussion will be postponed.

Currently, some manufacturers are specifying DAC components as having more than this error criterion. An inexpensive 16-bit DAC might have an absolute non-linearity specification of ± 2 LSBs. Such a converter is clearly not monotonic. Each level is very inaccurate. However, it does have the same accuracy as a 14-bit DAC component specified as ± 0.5 LSBs. If we had simply not used the lower two bits of this 16-bit converter, we would have reduced the error relative to the number of bits we are using. Manufacturers simply provide us the extra pins to control two extra bits. These two bits, although inaccurate, do provide some advantage. But again, we'll postpone this discussion, until later.

If you will reread the above discussion, you will note that we referred to the error as *absolute* non-linearity. This includes all errors from all sources: gain error, DC drift error, temperature error, etc. Many of these are of no interest to the audio engineer. They result from the fact that DAC manufacturers come to audio late and they are used to considering absolute errors. For the audio engineer, DC drift is almost always irrelevant. Similarly, a 0.1 dB gain change is not that important. And lastly, most audio equipment operates in a narrow temperature range of about 20 to 40 degrees Celsius. The only error which is of interest is *relative* nonlinearity: the degree to which the quantization levels deviate from a straight line. But only sometimes is this number provided. Fortunately, Murphy is working for us, since absolute nonlinearity is always worse than relative nonlinearity. On the other hand, manufacturers sometimes provide us only with "typical data," rather than worst-case. This means that the application engineer measured the first dozen units and then took the average. It says nothing about what we might expect in the current production run. You pay your money and take your chances. A maximum error means that each unit is, in theory, tested to be within that specification. To make life more interesting, there is a new phrase: "guaranteed of design." This means that the manufacturer intends the specification to be a maximum value, but he does not test each unit to insure that it works to within that specification. This is a more-conservative version of "typical." Whereas 50 percent of the units (or 100 percent) might be worse than a typical specification, only 10 percent might be worse than a guaranteed-by-design specification.

EFFECT OF DAC ERRORS

Let us consider an A/D encoder and D/A decoder made with an imperfect DAC, but with all other components as ideal elements. To do this analysis, we will use a trick. We will consider the equivalent analog error which corresponds to the incorrect digital word. This means that we will only consider the error in analog terms rather than in digital terms. The reason for this is that the quantizer may or may not result in an erroneous digital word. We used the same approach when we analyzed the theoretical error produced by a perfect quantizer.

Let us consider a few quantization levels located at −10 mV, + 10 mV, + 30 mV for a perfect converter. We defined the region between + 10 and + 30 mV as being + 20 mV. This results in some quantization error for an input signal such as + 12 mV. It is called + 20 mV in the digital domain (center of the quantization region) and there is a −8 mV quantization error. Now, let us introduce some error by allowing the "true" + 30 mV quantization level to be at an actual + 40 mV (+0.5 LSBs) and the "true" + 10 mV quantization level to be an actual 0 mV (−0.5 LSBs). Any analog voltage between 0 mV and + 40 mV will therefore be encoded to the digital word which is defined as + 20 mV. Notice that the maximum possible analog error is now ±20 mV, or ±1 LSB. So, depending on what assumption we make about the

Electro-Voice's Greg Silsby talks about the Sentry 100 studio monitor

Production Studio, WRBR-FM, South Bend, Indiana.

In all the years I spent in broadcast and related studio production work, my greatest frustration was the fact that no manufacturer of loudspeaker systems seemed to know or care enough about the real needs of broadcasters to design a sensible monitor speaker system that was also sensibly priced.

Moving to the other side of the console presented a unique opportunity to change that and E-V was more than willing to listen. When I first described to Electro-Voice engineers what I knew the Sentry 100 had to be, I felt like the proverbial "kid in a candy store." I told them that size was critical. Because working space in the broadcast environment is often limited, the Sentry 100 had to fit in a standard 19" rack, and it had to fit *from the front, not the back*. However, the mounting hardware had to be a separate item so that broadcasters who don't want to rack mount it won't have to pay for the mounting.

The Sentry 100 also had to be very efficient as well as very accurate. It had to be designed so it could be driven to sound pressure levels a rock 'n roll D.J. could be happy with by the low output available from a console's internal monitor amplifier.

In the next breath I told them the Sentry 100 had to have a tweeter that wouldn't go up in smoke the first time someone accidentally shifted into fast forward with the tape heads engaged and the monitor amp on. This meant high-frequency power handling capability on the order of five times that of conventional high frequency drivers.

Not only did it have to have a 3-dB-down point of 45 Hz, but the Sentry 100's response had to extend to 18,000 Hz with no more than a 3-dB variation.

And, since it's just not practical in the real world for the engineer to be directly on-axis of the tweeter, the Sentry 100 must have a uniform polar response. The engineer has to be able to hear exactly the same sound 30° off-axis as he does directly in front of the system.

Since I still had the floor, I decided to go all out and cover the nuisance items and other minor requirements that, when added together, amounted to a major improvement in functional monitor design. I wanted the Sentry 100 equipped with a high-frequency control that offered boost as well as cut, and it had to be mounted on the front of the loudspeaker where it not only could be seen but was accessible with the grille on or off.

I also didn't feel broadcasters should have to pay for form at the expense of function, so the walnut hi-fi cabinet was out. The Sentry 100 had to be attractive, but another furniture-styled cabinet with a fancy polyester or die-cut foam grille wasn't the answer to the broadcast industry's real needs.

And for a close I told E-V's engineers that a studio had to be able to purchase the Sentry 100 for essentially the same money as the current best-selling monitor system.

That was well over a year ago. Since that time I've spent many months listening critically to a parade of darn good prototypes, shaking my head and watching some of the world's best speaker engineers disappear back into the lab to tweak and tune. And, I spent a lot of time on airplanes heading for places like Los Angeles, Grand Rapids, Charlotte and New York City with black boxes under my arm testing our designs on the ears of broadcast engineers.

The year was both frustrating yet enjoyable, not just for me but for Ray Newman and the other E-V engineers who were working on this project. At this year's NAB show it all turned out to be worth it. The Sentry 100's official rollout was universally accepted, and the pair of Sentry 100's at the Electro-Voice booth was complemented by another 20 Sentry 100's used by other manufacturers exhibiting their own products at the show.

What it all boiled down to when I first started the project was that I knew that the Sentry 100's most important characteristic had to be *sonic integrity*. I knew that if I wasn't happy, you wouldn't be happy. I'm happy.

Greg Silsby

Market Development Manager,
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distribution of DAC errors, the actual quantization process may produce twice the error of the theoretical converter. As much as 1 bit of the converter is lost in terms of S/N or dynamic range.

A similar process happens at the D/A decoding process. If we again assume worst case, the +20 mV quantization level could be in the opposite direction of the encoding A/D. A 0 mV input signal could be encoded as +20 mV, and this level might be +40 mV at the decoding output. We now have a peak error possibility of 2 LSBs. Fortunately, this worst case does not happen for many of the quantization levels, and the random nature of the input means that the noise created is the average occurrence of such errors. To truly interpret the effect of DAC nonlinearities, we need to know something about the frequency of the errors. Is the worst-case specification only for one level? Is there a systematic pattern to the errors, or are they random? Unfortunately, no manufacturer will specify his DAC components in these terms, and we poor designers can only infer the pattern of errors based on our knowledge of the manufacturing process.

It would be very nice if we could assume that the errors were random. Then our understanding of the effects would be easy, since a random error would look like noise. Moreover, when a random error has a defined peak, we could expect the RMS value to be much smaller. But, sorry to say, most of the DAC component errors are not random, but systematic. To appreciate this, let us review the way these are built. Each bit controls a voltage or current which is added to the input. Let us consider the following unit. When all switches (bits) are off, the DAC produces −10.23 volts. The first bit (MSB-most significant bit) adds +10.24 if on, and nothing if off; the next bit adds +5.12 volts if on, and nothing if off; the third, 2.56, the fourth, 1.28, etc.

Using this example, we will examine the state of the switches as we go from −10 mV to +10 mV. To achieve +10 mV, the MSB is on and all other bits are off, since we must have the following sum:

10 mV = −10.23 (DAC baseline) + 10.24 (MSB) + no other switches.

The −10 mV case is rather different, since the MSB is off and all other switches are on:

−10 mV = −10.23 (DAC baseline) + 5.12 + 2.56 + 1.28 + 0.64 + 0.32 + 0.16 + 0.08 + 0.04 + 0.02.

Notice that to achieve a 20 mV difference, we are comparing two very large numbers, and one of these numbers is itself made up of nine numbers. The 20 mV difference is actually superimposed on a 10 volt pedestal. In contrast to this example, let us compare the case of the +10 mV and +30 mV levels. One case is, as before:

10 mV = −10.23 (DAC baseline) + 10.24

and the other is:

30 mV = −10.23 (DAC baseline) + 10.24 + 0.02.

To achieve this difference, we only need to turn on the LSB. If the LSB were in error by 10 percent, it would be 0.018 volts instead of 0.020 volts. The error seen is only 2 mV. However, a 1 percent error in the MSB would produce an error of 102.4 mV. In other words, a very small percentage error in one of the bits is likely to produce a large spacing between two quantization levels when the levels involve the changing of many bits. In the first example, all the bits were changing since the digital number went from 0111111111 to 1000000000, whereas in the second example only one bit changed. The digital number went from 1000000000 to 1000000001.

The first change is called a major carry since the sum of the LSB must be carried across all the bits. DAC errors are almost always concentrated at major carries. This is a far contrast from random.

The major carries, in audio terms, are at 0, +half scale, −half scale, +quarter scale, and −quarter scale. Except for the first, the major carries happen for high level signals. The carry at 0 volts is the most important in terms of audio—it is for very small signals—and it is the one which is likely to be largest. ■

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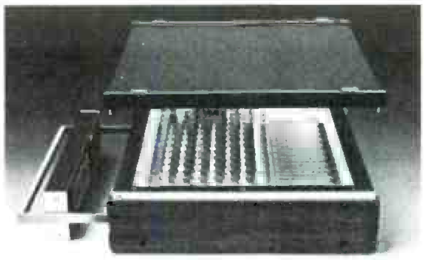

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Seeking a System

• Last month, we discussed the dimensions a digital audio system must encompass, based on information theory. Now, if we are to use that information, we must take a closer look at how to use it. A basic principle to keep in mind is that, since our objective is to produce an auditory illusion, we must consider how the human ear functions.

A fundamental difference is that we want to transmit, or record our audio program on a single channel or, on a very limited number of channels, while the human ear has a very complex nerve bundle to convey its received information to the recognition center in the brain.

We have a similar situation with video. Before the advent of color TV, a satisfactory system had been developed using sequential scanning; a dot of variable intensity that scans the whole picture area, line by line, 60 times a second or, to be more exact, 30 frames a second. Interlacing helped to reduce the apparent flicker, while improving definition by providing twice as many lines in a frame. The flicker frequency is raised from once every 30th of a second, which is the frame rate, to once every 60th of a second, which is how often the spot traverses the screen.

Then came color. That posed a system problem for which a number of solutions were proposed and tried. One of them was sequential scanning; to cover the picture area three times to get the full color picture, each time being like one color of a three-color print. This might work if all you look at is stationary objects, or pictures with very slow movement in them. But fast movement means that successive frames, in different colors, will be displaced, resulting in color separation when anything on the screen moves fast.

This was one deciding factor that led to the three-gun system that was finally adopted, so that the whole picture is covered in the single frame, interlaced scan, as before. That also had the advantage of compatibility; if a black and white

picture is transmitted, it is received as black and white, not a succession of color-changing pictures.

Now, how can we apply that to audio? Well, we have all those bits of information that we discussed last month to use in conveying the full audio information. How are we going to organize it? Can we use a frequency scan, which was one possibility we discussed?

What we are talking about now is a system that goes through the audio frequency spectrum once every five milliseconds (or a little faster, if we can spare the information), from 20 hertz to 20,000 hertz. However, we encounter a problem with that. You cannot sample a frequency of 20 hertz, plus or minus five cents, in five milliseconds, which would allow you only one tenth of a cycle of the 20 hertz frequency. And, of course, the time your scanner is on 20 hertz is only a very tiny fraction of that.

What all this means is that we must continuously sense all of the frequencies being scanned, picking off the information by scanning similar to the way that a TV camera tube works. Instead of scanning the image on the picture tube, you'd be scanning the outputs of a frequency analyzer, having however many frequencies we eventually decided were necessary. That would at least make it possible to get outputs from all the frequencies and code them into the transmission.

But now we need to think about something else. We picked 5 milliseconds (or less if possible) as a valid scanning time, because that was the time at which two events, repeated in quick succession, could be audibly identified as separate events. Without too much thought that seems sensible. But would you say that events separated by only, say one millisecond, would not be discernibly different from a single event? Just because you cannot separate them, does not mean that the presence of two instead of one doesn't make any difference.

If you doubt this, consider taking the same program, putting in a one millisecond time delay, and mixing the delayed signal with the undelayed signal. Will it make any difference? You know it will. As frequencies go in and out of phase, up the audio spectrum, you will get cancellations and summations every other 500 hertz. If the interval was a bit longer, it would sound like reverberation, but at that short interval, it will completely alter the character of the sound.

Unless you do that, you will not get that effect. Sequential scanning, in which you pick up each frequency only once during the scanning interval, will not produce that effect. But there is a related effect we need to consider.

Remember CBS Labs' so-called isophonic system? They showed that, if you reproduced the initial transients of musical sound from a string bass, for example, over little speakers placed to

enable you to get a stereo location on the sound, while the real low frequencies were reproduced from a woofer concealed under the sofa, our human hearing convinced us that the whole sound came from a location simulated by the little speakers.

We know that happens. But why? For one thing, it proves that in some way our hearing faculty pays much closer attention to transient sounds, such as those initial transients, than it does to "follow-through" sounds. If it is just a very sophisticated frequency analyzer whose output is fed digitally to the brain, how can it do that?

Ask yourself what information, that you know to be in the system (your human hearing system), could be used for that purpose? An obvious possibility is the AGC that we have already discussed at some length. An initial transient is invariably a sound that is louder than that being received immediately preceding it. This must involve the same nervous system that conveys the sounds heard. A sudden extra loudness sends a message to the brain, whether by the nerves associated with the acoustic transformer in the middle ear, or by the more detailed nervous system from the inner ear, which initiates a reflex action to "turn down the gain."

If you think about it, you realize that this also has to be the mechanism by which the ear can pay that special attention to transient sounds. And in turn, that must also provide the key to the ear's ability to separate sounds that occupy the same frequency spectrum.

Your ear is already receiving a melody and harmony that has been going on for some time, when a new instrument, or group of instruments, comes in with a new set of frequencies and overtones. But many of them are not new, taken individually. They are new only as a set, recognizable as whatever they are: brass, woodwind, percussion, or whatever. Your hearing pays special attention to this new sound, virtually ignoring the sound that was already there. You can still hear it, but it is not part of what your hearing pays special attention to.

It would seem obvious that the change in level that triggers your ear's AGC system is what enables your hearing to make this distinction. Now, that happens fast. To the best of our knowledge (we are open to correction), precise measurements on the speed of this response have not been made. But, assume they have, and suppose the reaction time is the same five milliseconds that has been identified as being necessary to hear repetitive sounds as separate events. What then?

If this is the "timer" that prompts a closer analysis of the change in sound five milliseconds after the change in level hits, then it will analyze the change in content at that instant. It can do so, because every frequency has a separate nerve fiber along which to transmit that information. Your digital system is not so equipped.

This would suggest that our system needs a device that similarly pays special attention to such transient sounds. A constantly-scanning device would not do it. When a transient arrives, some of the new elements would be "in" by this time around, while others would not make it until the next scan. For it to work right, all transients would have to be timed very precisely to coincide with the beginning of a frequency scan. Otherwise, the quality of the transient would be falsified.

There's another angle to think about, that may correlate with that. In the analyzer of the inner ear, vibration of the fluid couples with the resonant transverse fibers that stimulate the nerve endings. A soft tone, or a small level of some frequency, may stimulate only the nerve endings directly associated with that fiber. A louder one will stimulate not only more nerve endings associated with that fiber, but also some of those associated with adjoining fibers, whose resonant frequency is above and below the frequency doing the stimulating.

In the human ear, all of this information is assimilated to convey the intensity of that frequency element. It is that way, because nerve fibers convey information only relatively slowly, and so the system uses a lot of fibers to convey more information than each one can by itself. We need to translate that principle for use by a system with only one channel, but very short time-element bits.

Perhaps continuous scan is not the best. Maybe we need to be able to synchronize a scan with any transient that comes in, and perhaps fill in, between transients, with something more like a continuous scan. Another thing to think about is: do we need a complete scan from 20 hertz to 20,000 hertz, giving information about every frequency—however many we decide to use—in between, for every scan?

In video, continuous scanning seems to be necessary (who knows, maybe some day it won't be) for every point to keep its correct place in the picture. Sync signals are used to lock the scan for line and frame, and everything stays locked. But in video, you must have a complete picture. The frame must be filled with something. In audio, this is not necessarily true.

Maybe it would be better to devise a scan which accurately identifies each frequency whose intensity is enough to be significant, and then gives information about its intensity. Conceivably, this could reduce the information requirements a considerable degree. During each scan, each packet of information would consist of a code to indicate that the next data identifies frequency, followed by one that would separate frequency identification from intensity identification. Does this give you something to thing about? ■

N. I. WEINSTOCK

db Sound With Images

Scenes From a Video Notebook

• This observer's been doing a lot of reporting for the consumer press lately on the microcomputer revolution, "as it will have impact on both video and audio and join them together." Join them together? Well, these are the words one uses in describing pop phenomena in pop magazines—get more specific and the editor won't understand a thing, and will simply change it to his own even more misleading usage. (*Never!*—Ed.)

Consumers are not only getting the word, they're getting a lot of expectations. What these expectations mean for the audio professional is a changed industry. People are expecting to be able to afford a videodisc or VCR in their homes; the more affluent may find they must have both. This means at once greater opportunity—more programming surely will be needed—and greater quality expectations. The know-it-all final customer may say, "I can do that better at home," and then do it.

If the film experience is any guideline, certain standards—of syntax, of acceptability—will probably relax as video

becomes more and more available. As smaller, lighter film equipment became available in the 1960s—not only within that industry but to almost any interested party—jump cuts, uneven lighting, uncertain screen direction, jumpy cameras and other tics common to home movies, went big time. Expectations changed, and the language of film changed.

Now, many people are buying their own video decks. But how many of them will ever purchase *two* decks and the editing console necessary for smooth-looking cuts? I will go out on a limb and predict that glitches will become commonly seen, and eventually accepted as some form of scene transition.

Already out on the limb, I'll now go further. Recently, I was talking with a record producer who'd worked with Todd Rundgren on a recent album. One does not have to admire Mr. Rundgren's music to see some sense in his views of the home video medium—as this producer understands them, in any case. Rundgren is now involved in making video accompaniments to his (or others') music. Although he intends to make these as not merely accompaniments, he does not intend to produce much in the way of linear stories, either. Rather, in my acquaintance's expectations, a new *genre* is evolving, geared for giant-screen TVs that will constantly be playing as background—a cross between a wall hanging and a hi-fi. One puts on a record while talking with friends. Occasionally, something comes on that all will listen to; but often, as interesting as the record may be, it will merely be background. Deliberately, the record does not contain all "highs" (speaking experientially, not in terms of frequency). And, if it told a linear story that needed attention all the way through, it would probably be less successful. So will new video programming eventually turn out, my acquaintance and I agree. (It would be interesting to see if Mr. Rundgren agrees with this producer's interpretation of his work.)

Now, with a bit of a glitch and a different color balance to mark the passage of time, we switch scenes to a hundred degree (last August) press conference at which Technicolor announced their plans to market a seven-pound, quarter-inch videotape recorder. The recorder is made by Funai, of Osaka, Japan, and has been marketed there for several months already. By the time you read this, Technicolor supposedly will have tens of thousands of these things out in the retail stores, selling for $995.00. The unit employs quarter-inch video tape, in a cassette just slightly larger than a standard compact audio cassette. The tape is made by Fuji—only in half-hour lengths at first, then in hour-long size. The format is helical scan, with a loading system similar to that of VHS. Even if Technicolor does not come near their marketing objectives, I remain very impressed with this recorder—its quality may not be equal to half-inch, but it's not far from it.

Other quarter-inch formats are coming soon from Toshiba, Kodak, and, by 1985, Sony, so they say. In competition with all of this are portable half-inch VCRs coming in at lower prices than ever. Stripped down, non-portable units were shown at the Consumer Electronics Show by Sanyo and Sharp, bringing those prices down to the $600 range.

Where is all this going to take us? The Technicolor people expect their product to compete with Super-8 film, and not with larger-format video. Surely, Super-8 is already losing this competition. Eventually, quarter-inch video must also take over from half-inch in the home, just as the cassette has taken over from open reel in consumer audio. Someday, half-inch may be *the* professional medium. Or, more likely, it will linger, but constantly decline in importance, while three-quarter-inch solidifies its position. But the video "semi-pros" will use half-inch, as the audio semi-pros (what an awful word, says everyone who uses it).

With another glitch we slip back to the CES again, for two other developments of some interest to this column. The show could *almost* have been subtitled, Audio is Dead, Long Live Video.

I modified that with "almost," but, in retrospect, audio's descent was taken for granted by most of those at the show, and if video hasn't taken over everyone's affections, all saw it out there on the horizon. Among the variations of consumer adaptations being shown, were a couple with professional interest.

One of those products, that hopefully foreshadows the start of a trend that would be very welcome, was Akai's new, portable, VHS format, Activideo System. The important part: the soundtrack utilizes Dolby B, a step that ought to have been taken a long time ago. Easier provision for double-system recording (the usage of a separate audiotape recorder operating either with line-sync or connection-free crystal sync, such as those made by Kudelshi-Nagra, Tandberg, Uher, and others) would be the next logical step.

Actually, we were informed (and asked not to breathe details on this, so we proceed to speak in generalities) that a video noise reduction system is coming closer to being accepted by the manufacturers of video decks. Ray Dolby has been interested in an analogous video noise reduction system since his first patents. Such a system would be welcome indeed for home video, and in fact should be even less obtrusive than audio's Dolby B. But any more said would be hearsay.

The other promised development of interest in Chicago was Sony's new widescreen projection TV. Capable of reproducing in proper screen ratio a 70 mm or Cinemascope film that has been transferred to tape or taken straight off a film-chain, it is also capable of reproducing the stereophonic sound, with true high fidelity, that is often present in the large formats. For that matter, it's one of the few ways to get any high fidelity sound along with a video show, excluding systems custom-made or those that are much more expensive. It does not include its own speakers, but hooks up to external monitors. In all regards, including price, the unit hardly belonged at the CES at all, being perfectly suited for the professional who sets up systems (particularly impermanent systems) in industry.

Such a product should remain handy, as all of the video studio's gear becomes so portable that the studio itself will be all but obsolete. With the uses of video growing, and location work becoming the rule, the recording engineer will most likely be expected to be able to work with both audio and video. Yet, though the uses of the medium should keep growing, all signs point to its job glamour keeping the field as competitive as ever.

Enough of prognostications, for now. By next time we should have assimilated information from yet another show, the Fotokina in Cologne, Germany, and film will again take the stage. ■

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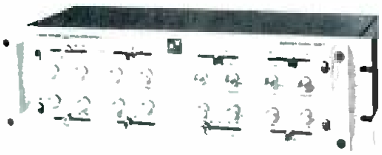

• The GEM-7 is an eight section full parametric stereo equalizer. It features four response shaping frequency banks per channel. Each channel has four frequency sections which are paired to adjust the 30-820 Hz and 820-16,000 Hz regions. Each frequency control bank is complemented with a gain control providing up to 18 dB of boost or attenuation as well as a variable bandwidth function for 0.16 to 2 octave range coverage. According to the manufacturer, the Bi-Fet circuitry incorporated by the GEM-7, is the first such circuitry to be offered to the audio market. Bi-Fets improve low noise, low distortion and high stability characteristics, while also enabling the increased gain range ± 18 dB. Full tape functions are provided for, with separate switching for record EQ, playback EQ and standard tape monitoring.
Mfr: Superex Electronics
Price: $449.95
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CARTRIDGE-HEADSHELL

• The M97HE-AH is a precision integrated cartridge-headshell with a universal four-pin bayonet headshell connector for instant installation in many turntables and tonearms. Its integrated design offers several advantages over separate headshells and cartridges, including easier installation, elimination of spurious resonances from insecure mountings and a total weight reduction of four to six grams when compared to many cartridge and separate headshell combinations. In addition, the M97HE-AH is provided with a special tonearm/cartridge alignment system which includes an overhang gauge and a non-operable alignment pin gauge stylus. It also features a nude-mounted hyperelliptical stylus, a viscous-damped dynamic stabilizer, telescoped stylus shank and a side-guard which protects against stylus damage. As a result, the M97HE-AH provides improved trackability in the mid and high frequencies at a tracking force of ¾ to 1½ grams.
Mfr: Shure Bros. Inc.
Price: $120.00
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BOARD SERVICE KIT

• Jensen Tools Inc. has developed a new kit for repairing printed circuit boards, with the emphasis on small component removal and replacement. Designated the JTK-47, the kit contains a 35-watt soldering iron, fork and hook soldering aids, a multi-position work holder and quick-adjust vise for holding circuit boards, DIP extractor, miniature chain nose plier and more. The tools are furnished in a 19 x 7 x 7-in. heavy-gauge steel tool box with a tray and extra room for additional tools.
Mfr: Jensen Tools Inc.
Circle 52 on Reader Service Card

COMPRESSOR/LIMITER

• The Gemini Easy Rider—a rack-mount, stereo/dual mono compressor/limiter, has recently been introduced by Audio and Design Ltd. Optimum attack time is calculated by a control which responds to program characteristics. Slower settings can be used safely, since the unit will adjust its attack automatically to handle unforeseen peaks. Dynamic attack change, relative to level, can range from 500 microseconds to five milliseconds. Release (recovery) time can also be programmed or set between 15 milliseconds and four seconds for specific signal shaping. The unit offers 33 dB make-up gain, with a 25 dB control range from limiting onset to a maximum clip level of +18 dBm, with preset output control user-calibrated between −10 dBm and +12 dBm referenced to limit threshold. A 20-segment LED bar graph, which reads gain reduction over the 20 dB scale, is set into the unit's laminated plastic front panel. Input/output/earth connections are via 12-way tag strip with independent "side-chain" access facilitated by a three-pole jack socket.
Mfr: Audio and Design Ltd.
Price: $875.00
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INTERCOM HEADPHONES

• Each TR-50 has a built-in crystal controlled FM transmitter, superhetrodyne receiver, standard nine volt battery supply and seven-inch receiving antenna. A limiting circuit prevents receiver overload, while a sensitive squelch circuit drives it into "quieting" during times of no transmission. Each operator hears his own side-tone as an indication that transmission is taking place. The wireless intercom headphones have five channels available for operation.
Mfr: R-Columbia Products Co., Inc.
Price: $275.00
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DUAL SIGNAL GATE

• A new audio processing device designated the SG-200 dual signal gate, was recently unveiled. The SG-200 contains two totally independent gates sharing a common power supply in one, 1¾-inch high, rack mount chassis. Operating controls for each gate include attack time, release time, range of attenuation, threshold, a totally silent in/out switch, and an internal/external gating command control.
Mfr: Symetrix Professional Audio Products
Price: $399.
Circle 58 on Reader Service Card

RECORDERS

• The 22-4 and 22-2 are two new recorder/reproducers in the Tascam Creative Series. The 22-4 is a compact 4-track 15 ips multichannel recorder with sync that features: function and output select; headphone monitor select; pitch control; optional dbx interface and optional remote pause controls. The 22-2, a compact 15 ips half-track recorder, features expanded scale VU meters, independent monitor and record ready controls, detachable head housing and optional remote pause control. Both units are three-motor three-head transports with precision moulded reel tables and spring-loaded reel holders.
Mfr: TEAC Corporation
Price: 22-4: $1,425.00, 22-2: $750.00
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AUDIO MULTIMETER

• The Bulgin Soundex Audio Multimeter is a multi-purpose instrument suitable for line testing and listening, peak program metering, amplification of microphone signals, calibration of peak program monitors, bench testing and other audio functions. The instrument combines a switched gain amplifier with 400v peak instrumentation input and a full spec Peak Program Meter capable of audio program level measurements down to -72 dB with 0.1 dB accuracy at center scale, as far as -50 dB. Amplifier input is fully protected to 400v, isolated and balanced to prevent grounding when connected to a jack field. The 50 ohm impedance output has sufficient power to drive headphones. Gain settings are achieved by eight pushbuttons on the front panel. Four other buttons provide On/Off, battery test, 600 ohm termination and access to a front panel variable gain potentiometer.
Mfr: H. R. Kirkland Company
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PERFORMANCE MIXER

• Incorporating today's technology in a live performance unit, the ARP8 features: monitor and effects submix bus; built-in analog delay to provide the desired echo effect; two 7-band graphic equalizers (one stereo for program left and right; one mono for the monitor submix) for balance among all the instruments in use; 3 bands of equalizers on each of the eight channels; 3 VU meters (for program left, right and submix), headphone, cue and talkback features; auxiliary inputs, direct bus inputs, stacking inputs, and effects send and receive.

Mfr: ARP Instruments, Inc.
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EQUALIZER

• The Model 3350 is a three range, 21 frequency, reciprocal 12 dB boost or attenuate equalizer. The high and low range equalization curves may independently be selected as either peaking or shelving. A 50 Hz to 15 KHz band-pass filter may be inserted exclusive of all other equalizer settings and an In-Out switch with LED status indicator silently switches the equalizer networks in or out of the circuit. The three frequency ranges are overlapping and are controlled by the outer knobs of concentric switches, the inner knobs of which set the amount of boost or cut in steps of 2, 4, 6, 9 and 12 dB. The Model 3350 is a panel mounting unit 5¼-in. high by 1½-in. wide and 5¾-in. behind the panel, requiring a bipolar 15 volt dc supply.

Mfr: Modular Audio Products
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DISCO SPEAKER SYSTEM

• A three-piece disco speaker system consisting of a low-frequency cabinet, a midrange/high-frequency cabinet and an electronic crossover/equalizer is now available from Electro-Voice. The HF12-3 speaker system features an EVM 12L woofer, E-v's VMR vented midrange speaker and an ST350A tweeter. The LF118 low-frequency speaker system is intended to be floor mounted, where it will produce undisturbed bass down to 28 Hz. Protecting the woofer from damaging subsonic information generated by record surface irregularities is one of the main features of the XEQ-1A crossover/equalizer. This is accomplished by an integral high-pass filter. Crossover frequencies are determined by plug-in modules, with the recommended crossover of this system being 125 Hz. The built-in switchable Thiele equalizer extends the response of the LF118 to 28 Hz. One XEQ-1A is required for each stereo channel.

Mfr: Electro-Voice
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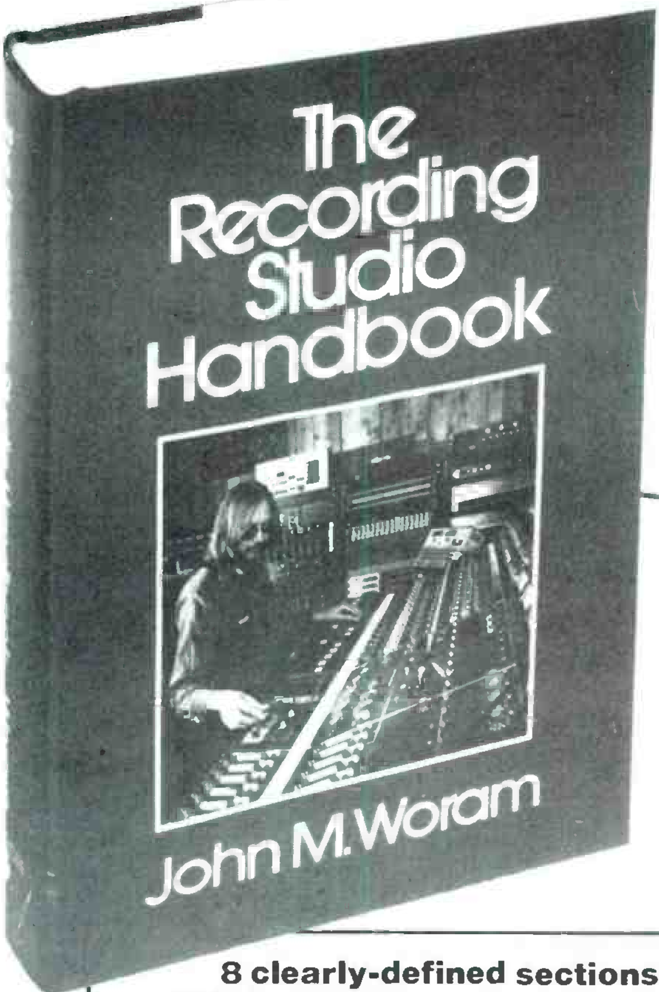

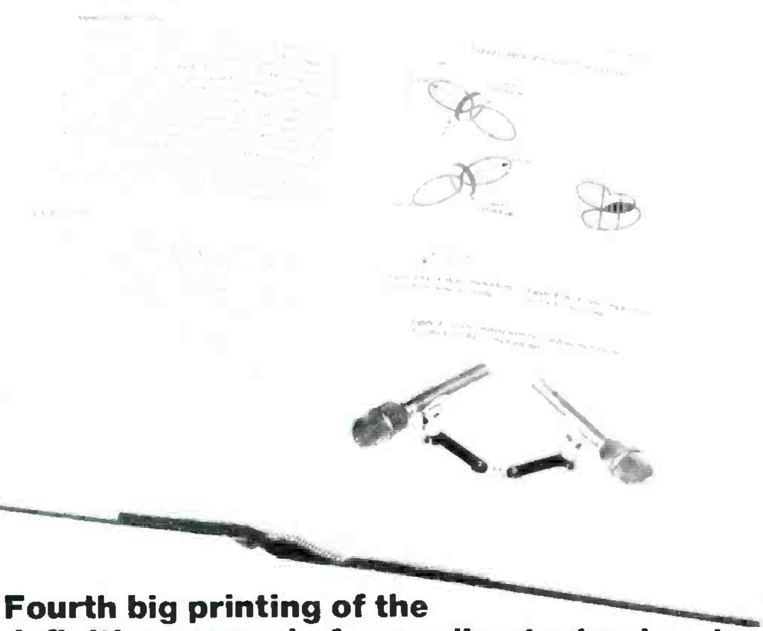

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db Editorial

"IN THE REAL WORLD, most—if not all—microphones are first-order." This little tidbit of news is from our microphone issue of July, 1978, and it refers to the types of polar patterns that are readily available in recording studio microphones.

Theoretically, the polar response of any microphone may be plotted from its "polar equation." For those who worry about such things, the general equation is, A + Bcos . As the values of A and B are varied, the microphone's pattern varies, from omni (A = 1, B = 0), through the various cardioid patterns, to bi-directional (A = 0, B = 1). In effect, the polar pattern potentiometer on some dual-diaphragm microphones varies the equation, to produce the desired response.

With a little knob-twiddling, we can set both A and B equal to 0.5, giving us the familiar cardioid pattern. As we continue turning, the cardioid pattern gets progressively narrower. At the same time, a rear lobe appears, and becomes progressively larger, until finally we arrive at a figure-8 pattern.

Now, let's just remove that rear lobe entirely, and we have a very narrow angle uni-directional mike: just the thing for miking an acoustic guitar surrounded by drums, bass and everything else. The mathematicians will tell us that to do this, all we need is a "higher-order" equation. For example, a sixth-order cardioid has extremely small rear lobes, and is down about 6 dB at only 25 degrees off-axis.

Hold on to your check books: here in the real world, most—if not all—studio microphones are "first-order," and you don't get your -6 dB until 70-to-90 degrees. As usual, we find that practice has a way to go before catching up with theory, and those higher-order mikes are still off in the future somewhere.

How far off? Some twenty years ago, the late Ben Bauer wrote, *"One feels we should be due for a 'breakthrough' in transducer technology."* He speculated about a *"...'zoom' microphone, in which the directional pattern can be adjusted to conform... to the optical angle of a television camera."* "A Century of Microphones" (Proceedings of the IRE, May 1962).

At the recent AES convention, engineers from JVC—the Victor Company of Japan—presented a paper entitled, "Zoom Microphone" (AES Preprint 1713, by Ishigaki, Yamamoto, Totsuka & Miyaji). The paper's conclusion: *"The zoom microphone... synchronized with video cameras, can provide a good integration of sound and picture."*

In other words, practice is catching up with theory, and the second-order concept, which was actually introduced by Dr. Harry Olson in the mid-40s, may eventually find its way into the real world. By the way, JVC's zoom microphone is made up of *three* spaced uni-directional elements, two of which are used to create the second-order characteristic. The microphone's polar pattern potentiometers are linked to the camera's lens, so that as the lens zooms, so does the microphone! (Just as Bauer predicted in 1962.)

Of course, there's a bit more to all of this than simply combining the outputs of two first-order microphones. Furthermore, the off-axis frequency response is still not as good as we would expect to see in a studio-quality microphone, although no doubt it's fine for its intended purpose. Of course, sooner or later we shall see (and hear) second-order microphones that *are* up to pro-studio standards. In the meantime, it pays to remember that when any two microphone outputs are combined, all sorts of unexpected patterns may result.

Last month's "mike quiz" (buried in our "Coming Next Month" paragraph) is now over. Of all the lures (for complimentary subscriptions) that we've planted in the pages of **db**, this one drew the least response, *and* the most wrong answers, too! (The right answer is to be found in this month's Application Note.) Perhaps this confirms what many of us suspect: microphones are often evaluated solely in terms of their "sound" (on guitar, snare drum, or whatever). Just as often, all the other variables are either misunderstood or ignored.

If this is so, we have an excellent excuse for a series of articles on microphones, which is just what you'll find in this month's **db**. We still have no word about our definitive kazoo mike, but after scanning these pages, perhaps you'll come up with your own definitives, which are better than ours anyway.

This month, our features range from theory, to practice, to anecdote. Bruce Bartlett reviews some of those little distractions that often get in the way of good microphone sounds. John Eargle offers some thoughts on miking pianos and other strings. And Ralph Hodges brings us some of engineer Fred Catero's views on mikes and miking (and everything else, for that matter).

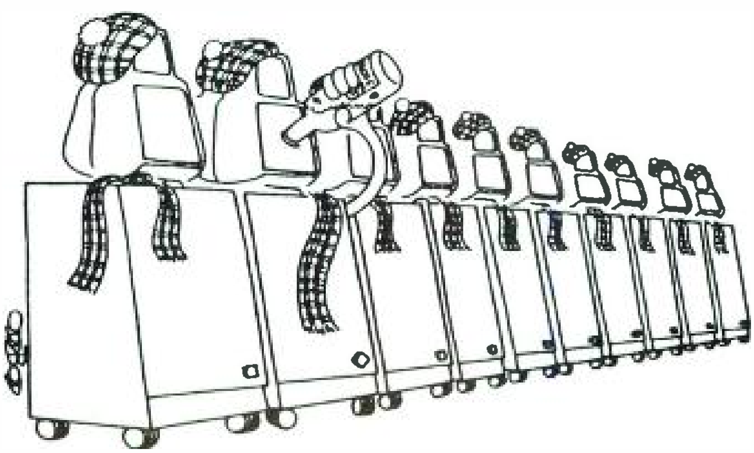
We also offer another **db** Application Note, and a directory of microphone manufacturers. And, since this is a transducer issue, that means loudspeakers too. We asked John Borwick to tell us about a two-mic, thirty-six speaker live performance/recording that recently tested the nerves of the BBC and KEF Electronics, Ltd. Also, Greg Silsby gives us an overview—from microphone to loudspeaker—of what it took to provide audio at the Republican National Convention.

For still more on the subject of microphones, keep an eye out for our early-1981 announcement of a new book written by John Eargle. The subject? Microphones, of course! No complimentary subscriptions if you spot the announcement, but there will be a special pre-publication offer to entice the early-birds in the audience. Keep watching! ■

JOHN BORWICK

Scene from Europe: A Classical Speaker Situation

The concert on the last night of the 34th Annual International Festival in Edinburgh, Scotland, presented the sound and television engineers with a unique problem.



ON SEPTEMBER 6th, a concert in Edinburgh's Usher Hall was given by the London Symphony Orchestra, conducted by Claudio Abbado. The work to be performed in the second half was the massive Berlioz *Te Deum*, in which the LSO would be joined by the Edinburgh Festival Chorus, the Scottish National Orchestra Junior Chorus, Philip Langridge (tenor) and Gillian Weir (organ). Actually, Gillian Weir did not exactly quite join the others—she played the recently-restored "Father Willis" organ in St. Mary's Cathedral, a mile away. The Usher Hall organ was in need of renovation and Claudio Abbado suggested the relaying of a distant organ. After thinking about Notre Dame in Paris and Paisley Abbey, it was decided that Gillian Weir would play the magnificent organ in St. Mary's Cathedral and the sounds would be relayed to the Usher Hall and reproduced as if the organ were really there in the Hall itself. This "place shifting" is quite common in the recording industry—we think of the recording of the Saint-Saens Third Symphony (CBS 2530 619) which brings together the Chicago Symphony Orchestra conducted by Daniel Barenboim in Chicago and Gaston Litaize playing the organ in Chartres Cathedral in France.

John Borwick is **db**'s *British correspondent.*

But the Edinburgh experiment was much more complicated and risky. The organ sounds had not only to be picked up accurately and transmitted to the concert hall, they had then to be reproduced at full volume and with a realistic spatial effect. There was also the need for complete and confident synchronism between the organist and the conductor with his 500 singers and musicians. Alan Bunting, the BBC's audio manager in Scotland, was in charge of all the technical arrangements for the programme and communications circuits. He also had to prepare everything for BBC Television, who were telerecording the Berlioz work live on Saturday for transmission by BBC 1 and several other countries on the following evening. He called in Raymond Cooke of KEF Electronics, and they devised an ambitious scheme using 36 of the KEF Model 105.2 loudspeakers, each pair powered by a separate Quad 405 power amplifier, 3,600 watts in total! (The model 105 is seen in *Made in Britain* in our June issue, and for a look at KEF Laboratories, see *If We Can Hear It, We Can Measure It* in the July **db**—Ed.)

Twenty of the speakers were placed in a straight line at the rear of the platform, along the base of the existing organ pipes, ten on each side of the organ console. A further group of sixteen speakers was situated centrally at the back of the Upper Circle to produce the antiphonal effects called for by Berlioz (though these were used at reduced power in the final balance). The BBC engineers used a simple crossed-cardioid pair of microphones to pick up the organ in the cathedral, placed at a rather close three metres, since the sound emerging from the speakers would have the added reverberation of the Usher Hall. The stereo signal from the microphones was modulated on to a radio transmitter/receiver system. This was also used to carry the two-way closed circuit TV signals, with cameras and monitor screens in both locations. Thus, Gillian Weir could see the conductor, and hear the orchestra and singers on a small loudspeaker placed alongside the console. Similarly, Claudio Abbado had a TV monitor on his rostrum which allowed him to see Miss Weir. He also had a microphone to talk to her during rehearsal, and I noticed that the assistant leader of the orchestra used this to ask Miss Weir to play an A to which the orchestra could tune before the leader and conductor made their entrances on to the platform.

The normal air of expectancy was made more electric as the musicians and the 2,500-strong audience asked each other "What happens if the organ link-up goes wrong?". The *Te Deum* opens with an interchange of loud chords between the orchestra and organ, and these are negotiated with no trouble at all. One of the main worries of the engineers had been to get a high-enough acoustic power level without overloading the amplifiers or speakers. The orchestra and organ chords both peaked to 90 dB on a sound level meter in the Grand Circle. The readings thereafter ranged from 37 dB(A) in quiet solos to 115 dB peaks when the five cymbals joined in. In fact, the organ seemed almost *too* loud at first to my ears, but there was no suggestion of clipping or distortion of any kind. During quiet passages from the organ, such as the extended introduction to the *Judex crederis* (which Berlioz described as "without doubt the most grandiose piece I have ever created"), the inevitable noises from the close microphone/organ balance did reveal themselves slightly. Using more distant microphones would have reduced this effect, but at the expense of picking up too much reverberant sound, not to mention the ambient noise, compounded of remote traffic, wind etc. As it was, I could detect the longer reverberation and just a suggestion of that indefinable cathedral ambience at each organ entry. Circuit

Figure 1. The scene in the Usher Hall, during a performance of Berlioz' Te Deum at the final concert of the Edinburgh International Festival 1980. KEF loudspeakers Model 105.2 are installed in two stacks of 10 at each side of the organ casing behind the choir.

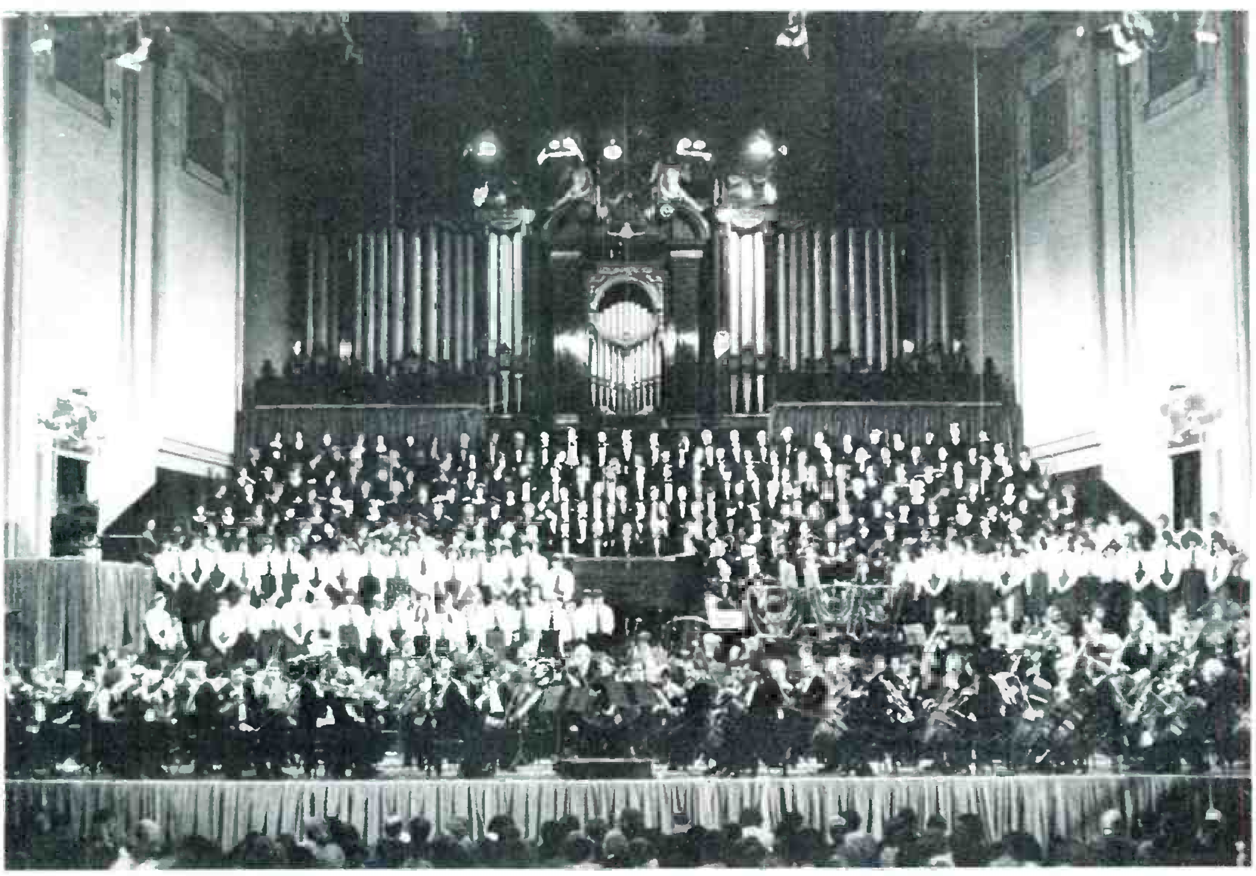


Figure 2. Laurie Fincham and Colin Munro of KEF Electronics setting up the loudspeakers in the Usher Hall.

Figure 3. The BBC mixing desk at St. Mary's Cathedral, Edinburgh.

noise in the radio links was also a contributing factor as it was only 53 dB below maximum signal level, and the organ dynamic range exceeded this.

The performance was voted a huge success by critics and audience alike. Will it ever be repeated this way? I doubt it. I spoke to Claudio Abbado and Gillian Weir afterwards and they both praised the technical and musical competence of the engineers concerned. However, they were looking forward to their next performance of the work a few days later in Belgium—where orchestra, choir and organ would all be in the same auditorium.

Such collaborations between musicians and engineers often arise in *avant garde* music of course (as well as in pop and rock concerts). I recently attended a concert of music by Karlheinz Stockhausen in which one work for live musicians and 4-track tape involved the composer himself seated at a Neve mixing console perched on top of the seats half-way back in the stalls—with huge speakers in all four corners of the Royal Festival Hall. On another occasion, I remember my tonmeister students at the University of Surrey being disappointed with the quality of a hired Steinway piano, which stood alongside our own Steinway in the University Hall for a concert recording of the Sonata for Two Pianos and Percussion by Bartok. Immediately after the concert, the students recorded it all over again—this time with our other excellent Steinway located in our music studio underneath the Concert Hall. They put TV cameras and monitors in both locations, with foldback headphones everywhere. And they proved their point. Comparing the recording made live at the actual recital with the new one—in which careful microphone balance had somehow matched the hall and studio acoustics, with judicious panning of the two pianos to quarter left and right—showed the superiority of our own instruments. And anyway, it was fun to do. ■

Figure 4. Two Calrec 1050C cardioid mics were used in a cross-cardioid configuration to mic the organ at St. Mary's.

Technical Data

ST. MARY'S CATHEDRAL

Organ microphones

Two Calrec 1050C cardioids, as angled coincident pair, 10 feet from pipes and 15 feet above floor.

Radio link

Main and standby picture transmissions at 7.125 gigaHertz with stereo organ signals on sub-carriers. Mono reserve channel on 141 MHz VHF.

Total of 14 audio circuits including two-way voice communication.

Circuit noise –53 dB.

USHER HALL

Local control

Two BBC 6-way mixing desks for stereo signal to front speakers and mono (left plus right) to rear speakers. Glen Sound distribution unit, 600-ohm balanced lines peaking to +10 dB.

KEF designed balanced-to-unbalanced converter/attenuator reducing level to 500 mV.

Amplifiers

18 Quad 405 stereo power amplifiers, one channel per speaker, delivering up to 28 volts rms (100 watts): flat down to 30 Hz, –2 dB at 16 Hz.

Total power 3,600 watts.

Total weight 162 kg (356.4 lb).

Loudspeakers

36 KEF Model 105.2 speakers, sensitivity 85 dB/watt at 1 metre.

Two bands of 10 speakers on each side of organ casing behind choir.

16 speakers at back centre of Upper Circle, grouped 6 + 6 + 4 on 22-inch risers.

Total weight 1,296 kg (2,851.2 lb).

Hall acoustics

Volume 565,000 cubic feet (compared with 2,500-6,000 cu. ft. for an average living-room).

Reverberation time 1.65 s (with audience).

Noise level less than 30 dB(A).

Measured music levels during performance 37 dB(A) 115 dB (linear).

Why The MTR-90

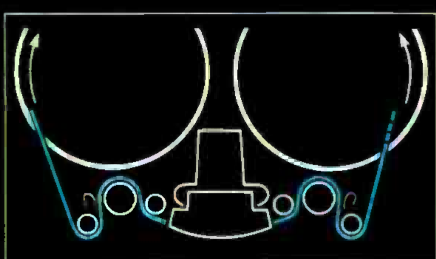
When you're buying studio time, it can save you a bundle. If you're selling the time to your clients it can make you a bundle.

It's the MTR-90. The 16/24 channel professional recorder that is the state of the analog art. It's the new machine that outpaces the big names. And, you know who they are.

Here's why we're so confident:

Superior Tape Handling

Especially critical for wide-width tape, the Otari Optimal Tape Guidance system was the industry's first three motor, pinchrollerless two-inch tape transport; a system so superior to conventional pinchroller designs that it is also utilized on a competitive machine costing twice as much



money. The MTR-90 treats the important ferric oxide tape coating like a precious metal. Compared to conventional designs, the sonic "shine" and brilliance of a master recording stays on the tape. Smooth, even tape packs in all operating modes are the rule, not the exception.

Award-winning recording engineer Phil Seretti, owner of the one-hundredth MTR-90 and producer Janja Vujovich

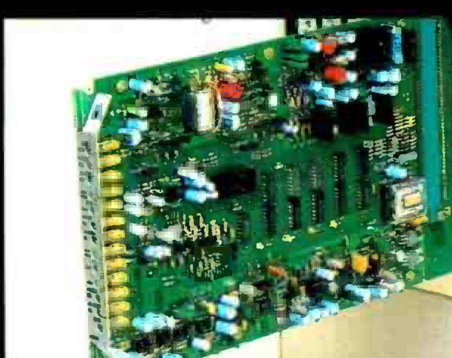
Advanced Audio and Control Circuitry

Easily accessible single card electronics not only save money, but reduce the complexity and problems of interconnection failures. Active mixing of audio and bias minimizes ringing and set-up difficulties. There's a transformerless playback amp for optimum transient response and dynamic range (greater than 71dB, 24-track: 30 ips, unweighted 30 Hz-18 kHz @ 1040 nWb/m). High slew-rate components in critical signal stages give you better aural results: Distortion-less than 0.5% at 1 kHz(250 nWb/m). Output: +28dBm.

Punch-ins and outs are totally transparent and effortless due to an integral digital timing section on each audio card that precisely ramps the erase and bias currents to yield "gapless" performance. Transport logic is digitally controlled for reliability and ease of servicing. There's a master crystal controlled reference clock for capstan, counter, record timing, bias and erase signals. Easy, rear-panel access to time-base functions facilitate SMPTE interface.

Easier To Maintain

The V.U. meter panel is hinged to give you wide open access. Remove the side panels, open the electronics bay doors and there is nothing you can't get to: power supply, master bias level, playback equalizations, motor drive amps, capstan and reel motors. The MTR-90 is designed for the real world of studios — where routine maintenance and care shouldn't have to be a headache.



Single card audio electronics contains record, reproduce, synchronous and timing circuits

Circle 26 on Reader Service Card

Outruns The Herd.

The Extra Thought

The MTR-90 makes sessions go smoother because we also designed-in such features as: a VSO that offers ±20% speed variation with 0.1% resolution; a precision, continuously variable edit and cue control; and whenever you go into start/stop or fast wind, a circuit automatically mutes playback level —

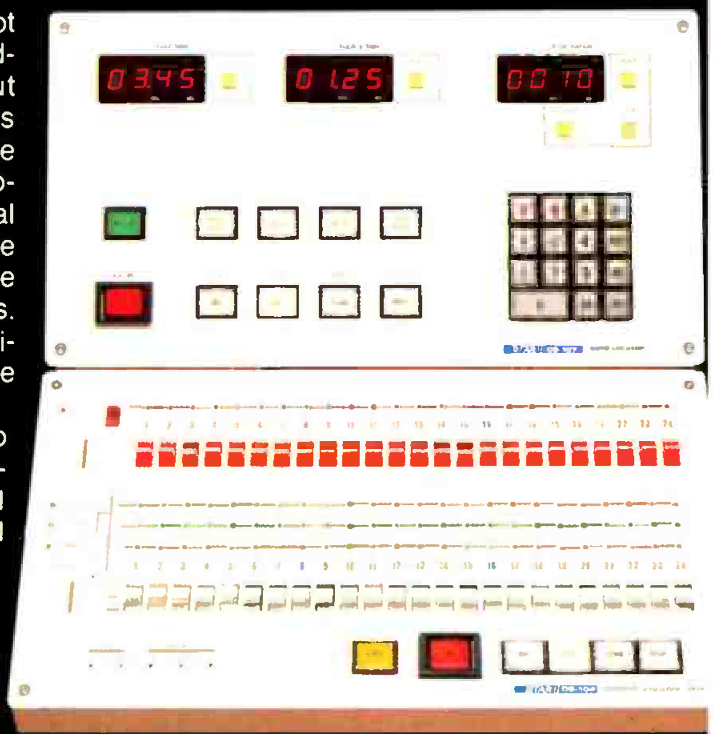
Easier maintenance with convenient access to all internal components

a feature that's designed to reduce ear fatigue and save your monitors. Every MTR-90 also comes with the industry's most advanced remote session controller. And, when you need the benefits of a sophisticated ten position memory locator, just plug-in to the back of the MTR-90 and place the optional locator atop the companion session controller's convenient roll-around pedestal.

If you've gotten the impression that it's advanced... good. If you also have any questions about how it will hold up to professional requirements, then let us assure you that we are prepared to stake our 16 year reputation for unparalleled reliability on the MTR-90. We've earned our place among audio professionals by competing with the best you've come to expect. Competing not only on features and advanced technology, but also with the ruggedness and essential service back-up so crucial to a professional product's total acceptance. After all, we do know that you make your living on our products. And we take that very seriously — because we make our living from you.

If you're moving up to a better, larger format machine, or moving the old one to the side, you need to get acquainted with the MTR-90. The New Workhorse. You'll find out for yourself why it outruns the herd. Just contact any of these fully committed dealers. Then arrange for a demonstration at your studios. The MTR-90 will give you every reason to consider that if you buy something else, you just might be buying something less... for more.

Watch for Otari's MTR-10 Series. They're the companion ¼" and ½" mastering machines that join the MTR-90 professional recorder.



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BRUCE BARTLETT

Hum, Pop, Thump and Other Microphone Noises

Presenting some techniques to aid in the reducing of unintended inputs and microphone noises.

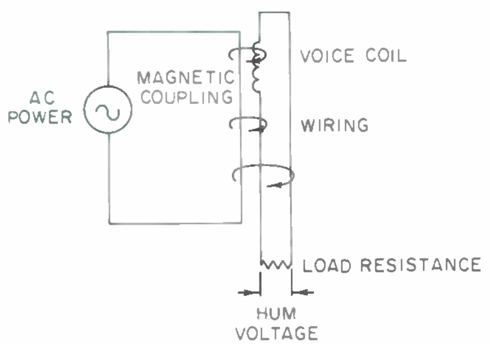
THE INTENDED FUNCTION of a microphone is to pick up sound and convert it to an analogous electrical waveform. Unfortunately, a microphone is also sensitive to unintended inputs, such as pop, hum, and handling noise. Wherever there is concern for high-quality audio, reducing pickup of these unwanted signals is important. To help the user do this, this article will discuss the nature and measurement of microphone-related noises, and will describe several methods to minimize them.

MAGNETIC HUM

In almost any room where a microphone is used, an alternating current exists in the electrical wiring inside the walls, floor and ceiling. The current oscillating through the wire conductors creates an oscillating magnetic field in the room. If a dynamic microphone is used in this room, the conductors of the microphone voice coil cut the magnetic lines of force from the oscillating field. An AC voltage is magnetically induced in the voice coil and, after amplification, this signal is heard as a low tone or buzz called "hum." Magnetic hum also can be induced in impedance-matching transformers and response-shaping networks with inductors inside the microphone. FIGURE 1 shows a schematic diagram of magnetic hum coupling.

Although AC power is commonly thought of as a 50 or 60 Hz sine wave, it is often rich in harmonics due to transformer core saturation. The higher the frequency of the oscillating magnetic field, the greater the rate of change of flux, and so the greater the induced voltage is. Thus, the high-frequency harmonics are picked up more readily than the 60 Hz component of the hum.

Figure 1. Magnetic hum coupling between a power line and a microphone.



Bruce Bartlett is a senior development engineer with Shure Bros.

High-frequency components are also much more audible. The result may be a "buzzy" sound in the hum pickup.

Other sources of magnetic hum besides power wiring are transformer radiation, SCR dimmers, and fluorescent lights. SCR dimmers operate by clipping the power-line voltage and, consequently, generate strong harmonics. Fluorescent light ballasts are reactive and radiate powerful hum harmonics magnetically.

A magnetic hum field and a microphone's hum pickup are both directional. Thus, a magnetic hum field can be detected by aiming a dynamic microphone in different directions while monitoring its output and noting a change in the hum level.

ELECTROSTATIC HUM

The voltage in the power wiring can also be electrostatically coupled to the microphone wiring; that is, a power wire and the microphone wire act as two plates of a capacitor. As before, the AC voltage induced in the microphone is heard as hum. FIGURE 2 is a schematic diagram of electrostatic hum coupling. With electrostatic hum fields, the high-frequency components are transmitted more easily through the capacitive reactance between the power wires and microphone wires.

The higher the microphone impedance, the greater the electrostatically-induced hum voltage. Thus, high-impedance microphones are more susceptible to electrostatic hum pickup than otherwise identical low-impedance microphones. (For more on hum pickup, see *Shielding and Grounding Revisited* in the October 1980 **db**—Ed.)

Hum pickup may be a problem whenever a microphone is used at some distance from a low-level sound source, since much amplification is required to obtain an adequate signal level. Any hum voltage induced in the microphone is also greatly amplified, resulting in a signal with audible hum. Unfortunately, several hum-reducing techniques are available.

Figure 2. Electrostatic hum coupling between a power line and a microphone.

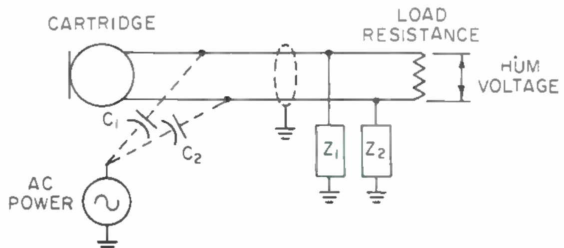


Figure 3. Synthesized pop disturbance, approaching and then striking the pop filter of a microphone.
Unlike sound waves, pop disturbances travel through the air by the mass movement of the air molecules themselves. A typical propagation velocity of the air mass is 15 feet-per-second. However, air particles within the turbulent mass may attain peak velocities of 60 to 90 feet-per-second. By comparison, a 100 Hz sound wave with a sound pressure level of 100 dB attains a peak particle velocity of only 0.03 feet-per-second.
Wind is also a moving mass of turbulent air and affects the microphone much like pop, except over a broader frequency range, and over a greater area.

1. *Magnetic shielding* utilizes a magnetically-conductive shield or case around the microphone cartridge, wiring and other internal devices. The shield offers a highly permeable path for the magnetic lines of force, conducting them around and away from the hum-sensitive microphone components. Grounding of this shield is unnecessary for magnetic hum reduction.
2. *Electrostatic shielding* employs a grounded electrically-conductive case or screen around the microphone cartridge, wiring and cable conductors. The shield offers a low-resistance path to ground for electrostatically-induced voltages.
3. *Twisted-pair wiring* inside the microphone and its cable enables the two conductors to occupy nearly the same position in space on the average. As a result, nearly equal magnetic hum voltages are induced in both conductors. These voltages cancel out at the load resistance when the balanced connection is used.
4. *With balanced lines,* nearly equal hum voltages are induced in each side of the line. Consequently, there is little differential hum voltage between the two amplifier input terminals to amplify.
5. A *humbucking coil* is often wired in series close to the dynamic microphone cartridge where it is exposed to nearly the same magnetic field as the voice coil. Since it is connected in opposite polarity to the voice coil, the humbucking coil generates an induced voltage approximately equal and opposite in polarity to that induced in the voice coil, thus cancelling hum by about 20 dB. Design effort must be directed toward achieving adequate cancellation at all frequencies. Humbucking construction of the microphone transformer helps it reject hum as well.

POP AND WIND DISTURBANCES

Another source of noise associated with microphones is "pop." When a person says words emphasizing "p," "b," or "t" sounds, a turbulent puff of air is forced from the mouth. If a microphone is placed within a few inches of the mouth, the puff strikes the microphone diaphragm and violently vibrates it, creating an electrical signal. Or, the puff may strike a surrounding grille structure and generate acoustical noise that is sensed by the microphone cartridge. The resultant explosive breath noise signal, when reproduced by a loudspeaker, sounds like a thump or little explosion called a "pop." FIGURE 3 shows a synthesized pop disturbance made visible by smoke.

In general, microphones have been observed to be most sensitive to pop at about three inches from the mouth. As the microphone is moved farther away, the intensity of the disturbance diminishes, because the pop loses energy with distance. Closer than three inches, the pop signal diminishes in intensity and loses low-frequency components because less turbulence is encountered and the disturbance acts over a comparatively smaller area.

REDUCING POP AND WIND SENSITIVITY

To minimize the sensitivity of a microphone to pop and wind disturbances, the disturbance must be weakened as much as possible before reaching the diaphragm by some sort of filtering structure or barrier. The filtering structure itself must generate very little acoustical noise when a pop hits it. A fine cloth or an open-cell foam-plastic screen in front of the cartridge commonly meets these requirements. These materials have a high resistance to high particle velocities (such as encountered in pop or wind disturbances), but have a low resistance to low particle velocities (such as encountered in sound waves). This non-linear filtering action reduces the pop and wind disturbances reaching the diaphrahm while passing acoustical signals.

The bigger the pop filter or windscreen, the better it works, but there are aesthetic and practical size limitations. Air-space size, foam thickness, and foam porosity are critical for optimum results. The user can further improve pop rejection by placing a foam screen over the existing microphone grille structure, keeping an air space, if possible, between the foam screen and the microphone cartridge. A typical example of a built-in pop filter is a ball-shaped grille such as shown in FIGURE 3. An external windscreen is shown in FIGURE 4.

Figure 4. An external windscreen.

Pop and wind noise can also be minimized by the use of omni-directional microphones. This type of microphone is approximately 15 dB less sensitive to pop and wind noise than a similar-sized uni-directional microphone. The net force of low-frequency sound waves acting on the diaphragm of a uni-directional microphone is about 15 dB less than that of an omni-directional microphone. To compensate for this lower operational force, the diaphragm damping of a uni-directional microphone is made fairly low to increase low frequency electrical output. Unfortunately, this also increases pop and wind sensitivity. Thus, an omni picks up much less pop than a uni. Note also that bi-directional ribbon microphones are extremely sensitive to pop and wind and, therefore, require very effective windscreens.

Pop disturbances are emitted from the mouth within a narrow conical angle. The user can take advantage of this fact by placing the microphone out-of-the-way of the air puff at the corner, above, or below the mouth. Note, however, that "t" sounds are directed downward. The reader can experience these effects by saying "p" and "t" sounds at his hand at various distances.

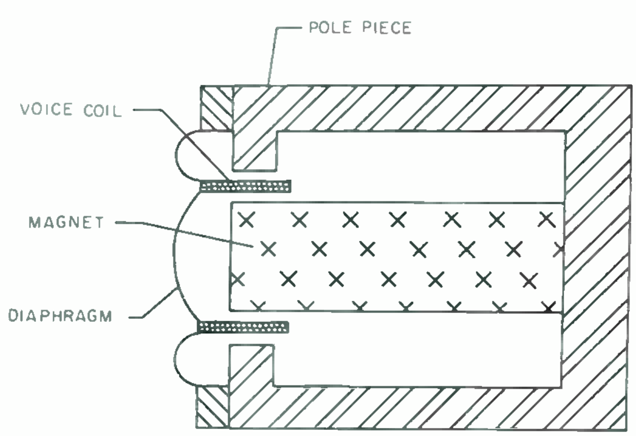
The spectral amplitude of pop signals is strongest at low frequencies (usually around 100 Hz) and decreases as frequency increases. Thus, some pop reduction may be attained by rolling off low frequencies on a mixing console when "p" sounds occur in the vocalist's signal.

PICKUP OF MECHANICAL VIBRATION

Microphones are often used in situations where they may touch sources of mechanical vibration or shock. Some such situations are: handling noises during hand-held use; stand noises associated with floor, boom, or desk-mounted placement; clothing noises due to lavalier use, and cable noises occurring with all applications. Much of this shock excitation covers a wide frequency spectrum.

How do mechanical vibrations cause an output from a microphone? Consider a moving-coil microphone. It consists of a diaphragm with an attached voice coil that is suspended in a narrow circular gap in a magnetic structure (FIGURE 5). A strong magnetic field exists in the gap. Relative motion between the voice coil and the magnetic gap induces a voltage in the voice coil, producing an electrical output from the microphone. This relative motion can be caused in two ways: by sound waves vibrating the diaphragm and voice coil with respect to the magnetic structure, or by mechanical excitation vibrating the magnetic structure with respect to the diaphragm and voice coil. Due to the inertia of the diaphragm/coil assembly, it tends to remain motionless when the supporting magnetic structure is suddenly vibrated by a shock. Consequently, there is relative motion between the voice coil and magnetic structure, which produces an electrical signal heard as a thump or scraping-handle sound. Microphones are most sensitive to shock in the direction of normal (axial) diaphragm motion.

Figure 5. Sectional view of a dynamic microphone.



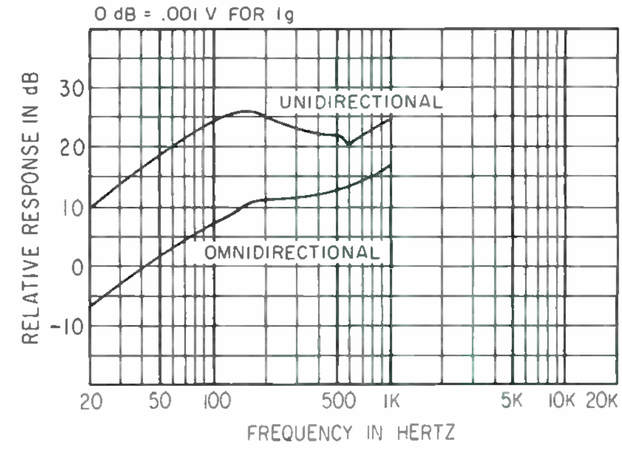


Figure 6. Vibration sensitivity of microphone cartridges.

A condenser or ribbon microphone has a diaphragm of much lower moving mass than that of a moving coil microphone. Thus, the condenser or ribbon microphone is inherently less sensitive to structure-borne noise than a moving coil microphone of similar size and directional characteristics.

VIBRATION REDUCTION TECHNIQUES

One goal in designing a microphone is to reduce the undesired output caused by mechanical vibration; that is, to minimize the diaphragm motion due to a given mechanical excitation.

In a dynamic microphone, the diaphragm response to shock is greatest at its resonance frequency (typically around 150 Hz in a uni-directional microphone, or 500 to 1,000 Hz in an omni-directional unit). The response to shock at resonance can be reduced by increasing the diaphragm damping resistance. This damping is partly provided by the internal friction of the diaphragm itself and mainly by acoustical damping material placed behind the diaphragm. There are limits to the amount of damping that can be used, since damping also can affect frequency response and directional characteristics.

As stated before, the diaphragm resonance of an omni-directional microphone is much more damped than that of a uni-directional microphone. Consequently, omni-directional microphones are approximately 15 dB less sensitive to mechanical vibration than uni-directional microphones of comparable frequency response and transducing principle (see FIGURE 6).

A popular technique of vibration reduction uses a high-compliance internal shock mount. In this method, an elastic mounting is used between the cartridge and microphone case to

Figure 7. Graphical demonstration of the influence of a shock mount on the vibrational sensitivity of a dynamic microphone cartridge.

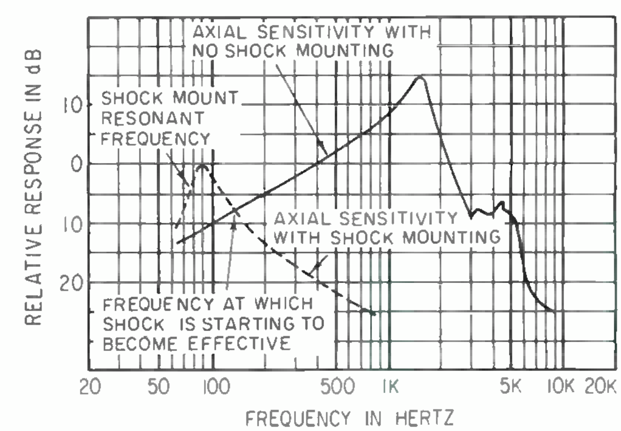


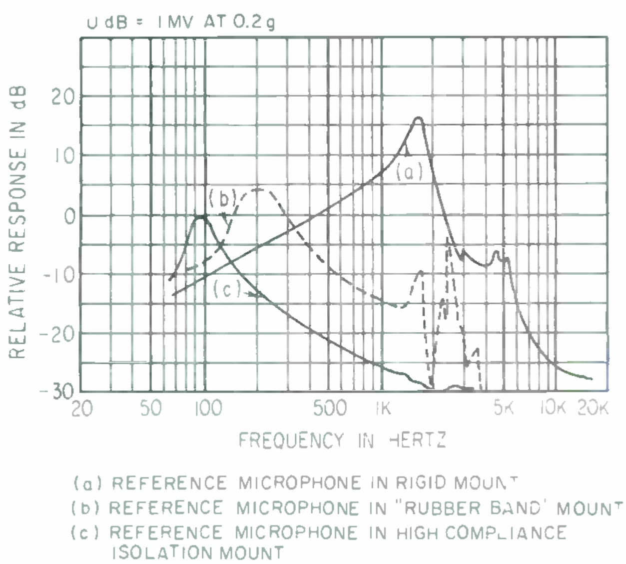
Figure 8. A high-compliance shock mount.

isolate the cartridge from vibration. The mass of the microphone cartridge and the compliance of the elastic suspension form a mechanical system with a certain resonance frequency. Mechanical excitation at this resonance frequency results in large motion of the microphone cartridge. The resulting output from the cartridge will be greater in this case than if there were no shock mount at all. However, at frequencies above resonance, the excitation transmitted to the cartridge becomes less as frequency increases, because the mass reactance of the system increases with frequency. Thus, at some frequency above resonance, the shock-mounted cartridge is producing less output voltage than it would be if it were not shock-mounted. In short, the shock-mount acts like a partially damped low-pass filter to mechanical vibrations transmitted from the microphone case to the cartridge (FIGURE 7).

By making the shock mount sufficiently compliant, the shock-mount resonance frequency can be placed below the lowest acoustical frequency of importance for the microphone. There are limits to making the shock mount too loose or springy; the cartridge may "bottom out" when it vibrates, especially since microphone size usually is limited. Maximum compliance should be in the axial direction because that is the direction in which the cartridge is most sensitive to shock.

For a further improvement in vibration isolation, the entire microphone can be suspended in a compliant shock mount placed on a microphone stand. "Rubber-band" type shock mounts can provide sufficient compliance, but are generally large and obtrusive. A device such as the one shown in FIGURE 8 provides effective shock isolation in a small package. FIGURE 9 shows its performance. It has maximum compliance in the axial direction where it is needed most.

Figure 9. Comparison of microphone performance with various types of shock mounting. (A) Rigid mount. (B) "Rubber band" mount. (C) High-compliance isolation mount.



MICROPHONE SELF-NOISE

Microphones and their associated preamplifiers produce minute noise or "hiss"-like signals. The major source of noise associated with dynamic microphones is the thermal noise of the resistance part of the microphone impedance. The subjective noise level of these microphone types decreases as the microphone sensitivity increases for a given impedance level.

Professional-quality condenser microphones generally have a lower subjective noise level than dynamics. With condenser microphones, noise is produced mainly by the impedance-conversion circuitry within the microphone itself. Some microphones are quieter than others, depending on the active-to-stray-capacitance ratio, the cartridge output level and the input resistance of the impedance-conversion circuitry.

RECOMMENDATIONS

The following is a summary of recommendations for reducing undesirable microphone noises:

For minimum hum:
- Use condenser microphones with adequate shielding or dynamic microphones with humbucking coils.
- For microphones without humbucking coils, use those with metal cases that provide adequate shielding.
- Avoid SCR dimmers and fluorescent lights.
- Orient the microphone for minimum hum pickup.
- Use balanced lines.
- Use low-impedance microphones.

For minimum pop:
- Choose omni-directional over directional microphones, if feedback and background sounds are not severe.
- Use microphones with built-in filters.
- Place a foam windscreen over the microphone grille, keeping an air space between the foam screen and the microphone cartridge.
- Use the microphone closer or further than three inches from the mouth.
- Place the microphone out of the path of pop travel.

For minimum pickup of mechanical vibration:
- Choose omni-directional over directional microphones, if feedback and background sounds are not severe.
- Generally, choose condenser microphones over ribbons, and ribbons over moving-coil microphones. Note, however, that some shock-mounted moving coil microphones are comparable in vibration isolation to condenser and ribbon microphones, depending on the effectiveness of the shock mount design.
- Place microphones in shock mounts on stands.

For minimum self-noise:
- Choose high-sensitivity dynamic condenser microphones designed for low-noise performance. ■

ACKNOWLEDGEMENT

The author wishes to thank R. Anderson, D. Arnold, W. Bevan, T. Locke, G. Plice and R. Schulein for their contributions to this article.

References

1. L. J. Anderson, "Sensitivity of Microphones to Stray Magnetic Fields," *Transactions of the IRE-PGA*, pp. 1-6, January February 1953.

2. Robert B. Schulein, Charles E. Seeler, and William R. Beven, "Design of a Studio-Quality Condenser Microphone Using Electret Technology," *Journal of the Audio Engineering Society*, Vol. 26, No. 12, pp. 947-957, December 1978.

3. Terry R. Locke, "An Effective Mechanopneumatic Shock Mount for a Dynamic Microphone," *Journal of the Audio Engineering Society*, Vol. 26, No. 9, pp. 623-628, September 1978.

4. Gerald W. Plice, "Microphone Accessory Shock Mount for Stand or Boom Use," *Journal of the Audio Engineering Society*, Vol. 19, No. 2, pp. 131-137, February 1971.

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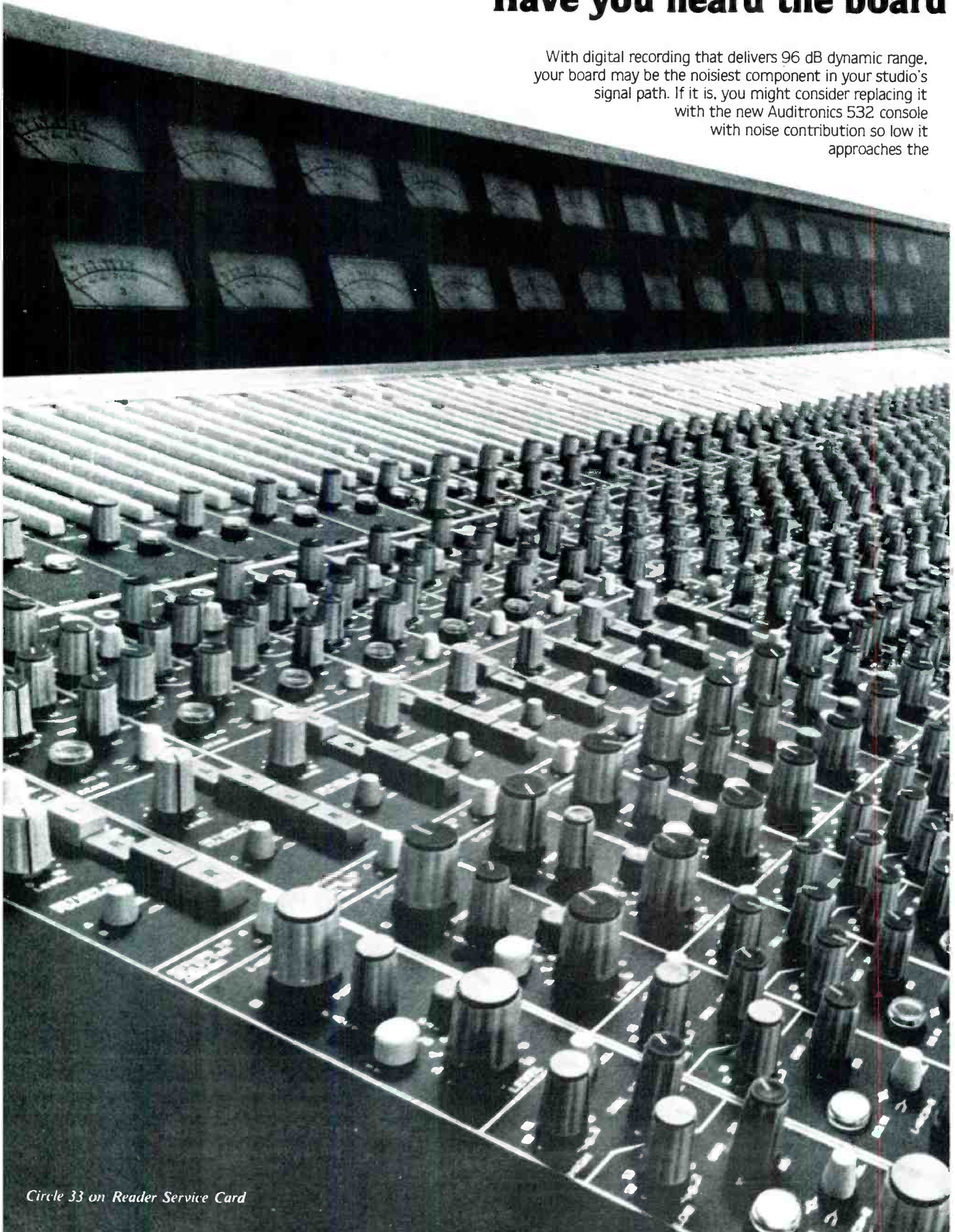

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JOHN M. EARGLE

Studio Microphone Techniques

A few suggestions for miking pianos and other instruments with strings, excerpted from the author's forthcoming microphone textbook.

THE MERE MENTION of studio microphone techniques evokes a wide range of responses from recording and broadcast engineers. Many have evolved their own unique techniques and problem-solving procedures through years of "cut-and-try" experiments, while others have spent their early years assisting and studying more experienced engineers.

Here, we'll offer some "textbook" solutions to common studio problems, with much of the advice coming as a direct application of microphone basic theory. There is always room for other approaches, and the enterprising engineer will keep a good eye and ear out for what others are doing, and for the many techniques which are being developed in studios everywhere.

KEYBOARD INSTRUMENTS

No other instrument poses quite as many pickup problems as does the piano. The sound radiation pattern is complex, and the spectral content may vary considerably with even the slightest re-positioning of the microphone(s). For example, consider the brief musical passage shown in FIGURE 1. Several single-microphone placements are also shown, along with the one-third octave peak-hold spectrum for each position.

CHOICE OF PIANO

The studio trying to make do with anything less than a six-foot grand is making a mistake. Better yet are the seven-foot grands—and of course, a nine-foot grand is pure luxury! The bass registers of smaller instruments tend to produce a wooden sound, no matter how well-regulated they may be.

Conventional wisdom states that pianos improve with age. However, this is not true. A well-built instrument reaches its prime in a comparatively short time, after which it begins a slow period of deterioration. The fact that many newer pianos are not built as well as their predecessors may suggest that the instrument will probably "mature" over the years. But this never happens. A well-regulated new instrument will exhibit even voicing, and just as important, mechanical noises due to the action and dampers will be minimal.

CHOICE OF MICROPHONE PLACEMENT

Some suggested microphone positions are shown in FIGURE 2. The stereo placements are intended for pickup of the solo instrument, and will produce a broad spectrum of sound. A single microphone will generally produce the best mono pickup if it can be located at some distance from the instrument. Alternatively, the stereo pairs may be combined for mono. In doing this, note carefully any apparent cancellations in the spectrum. An evenly-played chromatic spectrum through the mid-range will usually identify this.

It should never be necessary to place a microphone under the instrument. Not only is the sound extremely dull, but the thumping noise of the damper action may become obvious.

When the piano is to be picked up in a crowded studio, where the sound level from other instruments is quite high, the techniques shown in FIGURE 3 may be the only ones that will work. A microphone placed in any one of the holes in the cast frame will produce a distinctly unnatural sound, but one which may be quite workable—or even preferable—in pop or rock recording. Omni-directional microphones will be best. If the sound is too "mid-rangy," try adding another microphone at one end of the holes toward the tail of the instrument. If more high end is desired, a microphone may be added above the metal frame close to the top 1½ octaves of strings. With microphone one doing most of the work, microphones two and three should be carefully mixed in, to provide the desired spectral correction. This is, of course, a *mono* pickup, with the outputs of all three microphones combined.

If even more isolation from loud instruments is needed, a heavy blanket may be draped over the piano lid, with the lid at "half-mast."

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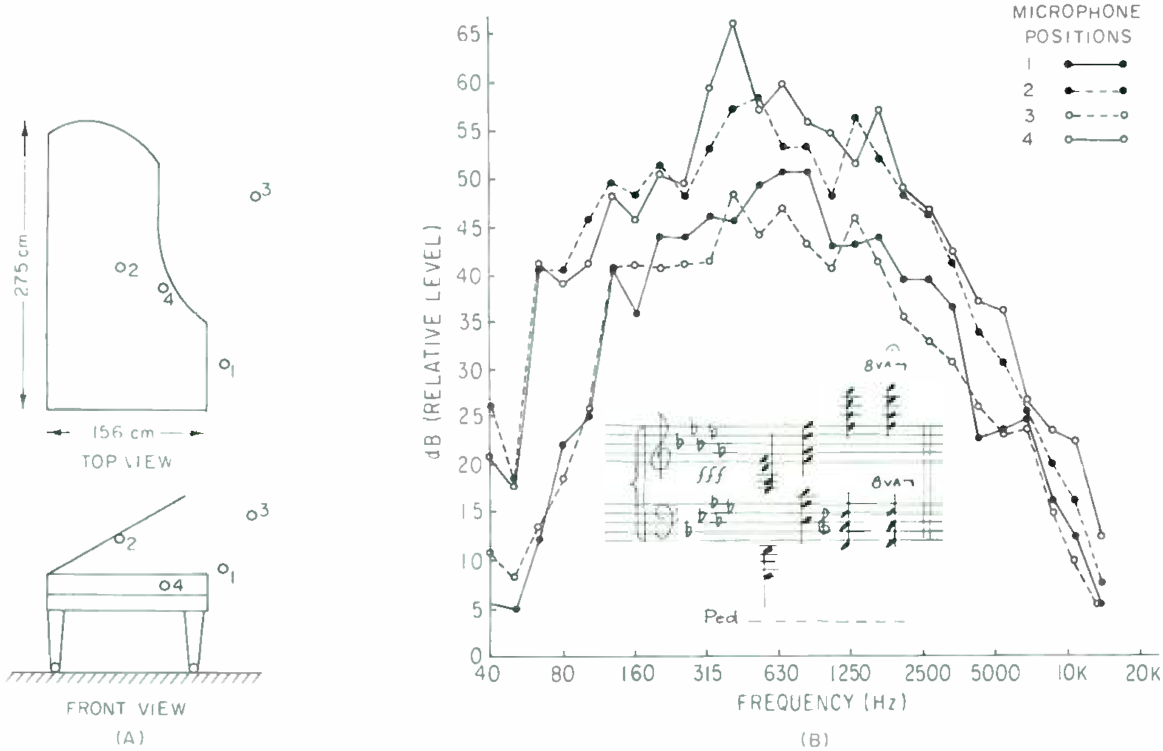


Figure 1. Variation of one-third octave peak-hold spectra for various microphone positions in piano pickup. (A) Top and front views, showing four microphone placements. (B) The brief musical passage produces different spectra at each microphone position.

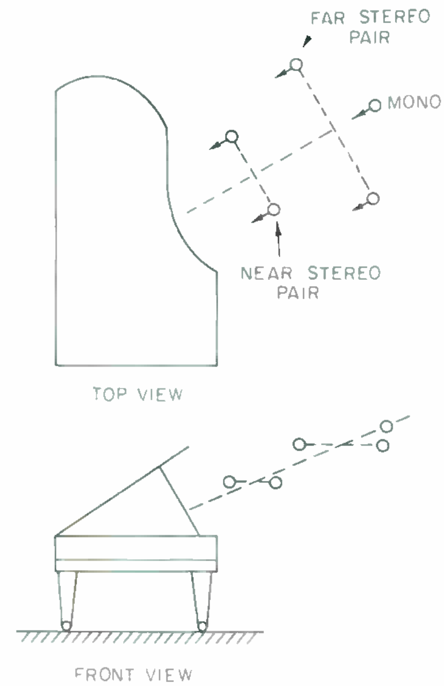
The techniques described above should result in a sound well-isolated from anything else in the studio. It will be quite dead, and accordingly may benefit from a slight touch of artificial reverberation. In-line equalizers, especially if they are capable of band-reject action, may help to tone down the brightness of the sound somewhat.

Another important observation of this kind of pickup is the effect of normal dynamic range changes in the playing of the instrument. The piano is capable of an extremely wide dynamic range, and loud passages will always be brighter than softer ones. They may, in fact, become too bright for such close pickup. In that case, the performer should be requested to contain the dynamics within some allowable range determined experimentally. Any additional dynamic control can always be provided by the engineer and producer in the control room. *Never* wait until the time of the session setup to determine the pickup of the piano. It is, after all, one instrument that is always in the studio, or will have been brought in well ahead of the session. Thus, there should be ample time for experimenting. Do not be afraid to tell a pianist to alter playing style, as may be required by the needs of the session. Some players are noisy pedalers, and close microphone placement may pick up too much thumping noise from the dampers. A slight change in technique on the part of the player will usually fix this.

CLASSICAL PIANO

For classical piano, the normal studio environment will not be satisfactory. There are only a handful of really good studios around the world suited for this purpose. The best recording venues are medium-sized live recital halls, or old-fashioned ballrooms. What is most desireable is a room whose reverberation characteristics emphasize the mid- and high-frequency portions of the spectrum. Many halls which are excellent for orchestral recording may simply be too big for piano recording.

Figure 2. Some suggested microphone placements for normal stereo pickups.



A fairly high level of reverberation may be perfectly satisfactory, if it is not too long. Like speech, the piano needs to be well-articulated, and a too-long reverberation time will blur musical detail. It should not exceed about 1.5 seconds. Some details of microphone setup are shown in FIGURE 4.

Many piano recordings suffer from poor focus or imaging in stereo playback. A coincident or quasi-coincident approach will cure this. If a spaced-apart approach is favored, there should *always* be a center microphone to stabilize the image.

Establishing the fore-aft position of microphones is critical. It is very instructive to begin close in and move the microphones outward in *small* steps, perhaps no more than about 30 centimeters at a time. There is often a fairly-narrow range, depending on the room and the instrument, where all musical details seem to fall into place. It is well worth taking the time to find the choice location for the microphone. Especially in classical recording, one cannot stress the importance of a fine instrument enough. Nothing less than a nine-foot instrument should be used, and it should *be in tune*.

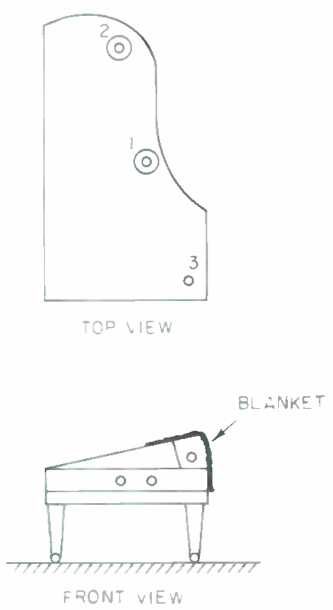
HARPSICHORD

Two other keyboard instruments are notorious for their noisy actions: the harpsichord and the celesta. These instruments are encountered less and less in pop recording, since their specific roles have been pretty much filled by various electronic keyboard instruments.

The harpsichord has the noisiest action of just about any instrument. When a note is released, the plectrum contacts the still-vibrating string before the damper can mute it. In the mid and upper portions of the keyboard, this may not be a problem, but the lower keyboard suffers from various "clunks" and "thunks" of the action. At close quarters, one is aware all the more of the action noise, as well as a high-end "sizzle." Clearly, the harpsichord is an instrument meant for a specific environment: it works best in intimate halls with reverberation times up to say, two seconds. Such environments tend to downplay the raucous high end considerably.

The engineer who is faced with recording a harpsichord as part of a typical pop session has few options. The instrument produces only a moderate level, and is inherently of limited dynamic range. A single omni-directional microphone located close to the raised lid, about one-third down the instrument, will pick up a good balance with minimum action noise. As a rule, its use in pop recording will be relegated to passages where it is exposed and not forced to compete with louder musical resources.

Figure 3. A closeup mono pickup of the piano.



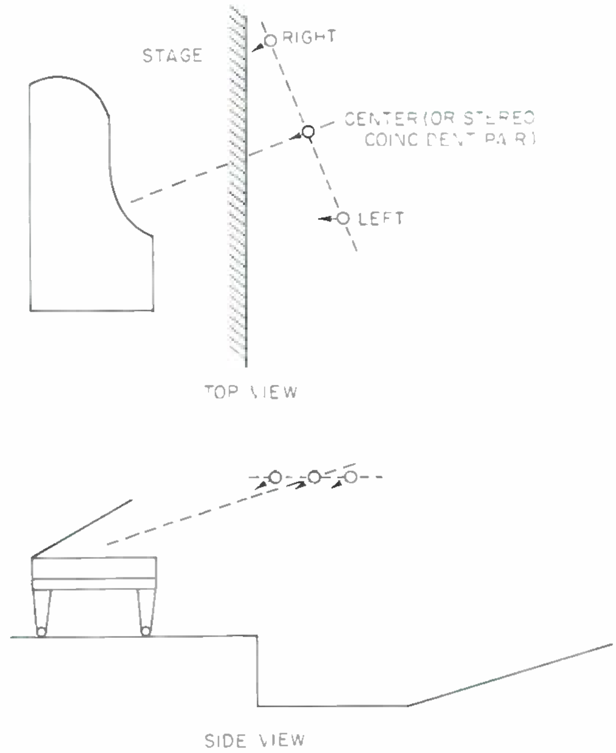


Figure 4. Recording the piano in a concert hall.

STRINGED INSTRUMENTS

Bowed instruments possess a fairly-wide dynamic range, but their overall output capabilities are limited. This, of course, is why they are used in large groups. In most pop recording, the string choir, consisting of violins, violas, cellos and string basses, is used for adding warmth, or for "sweetening" the musical texture in the form of a counter melody or other secondary musical role. Because of this, they are often over-dubbed at a later date, when the studio is free for them alone. A single stringed instrument may be picked up monophonically if that will satisfy the musical requirements. In this case, a single microphone located about 1 or 1.5 meters above the sounding board will suffice. However, where an ensemble is involved, other techniques will be required.

Even though the session budget may demand that you make an attempt, it is difficult to make a handful of strings sound like a large group. Microphones should not be too close, since that will invariably produce a harsh sound. A natural stereo spread to the string ensemble sound is essential, and multi-mono pickups are usually disastrous. While coincident pickup will produce an integral, well-spaced sound, the particular goals of pop recording will probably be met best through the use of spaced-apart microphones, with sufficient acoustical leakage between them to add the required warmth. In fact, the spaced-apart approach will make the group seem somewhat larger than it actually is.

STUDIO SETUP

Allow sufficient room for the string players. Normally, they are used in pairs, with each pair requiring about 1.5 meters on a side. Microphones should be located overhead, at a distance of about three meters. The number of microphones will depend on the size of the group. A typical group may be made up of six first violins, six second violins, four violas, four cellos and one or two basses. A group of this size may require up to six or seven microphones, each panned across the stereo array as required. An additional microphone may be needed for the string basses to provide a firm bass line under the control of the mixer and producer.

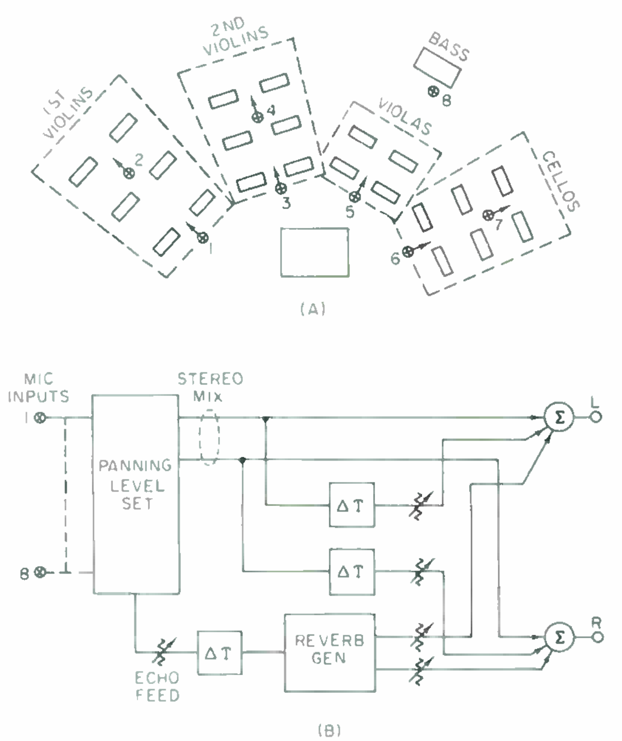


Figure 5. Signal processing scheme for studio pickup of a medium-sized string ensemble. (A) The floor plan. (B) The signal flow diagram.

REVERBERATION

Artificial reverberation is usually a necessity in recording a small group of strings in even a large studio. While some studios can be adjusted to be fairly reflective acoustically, they are usually too small to produce a significant reverberant field so essential to good string sound. Today's better stereo reverberation generators, both digital and analog, are equal to the task. A typical floor plan and signal flow diagram for recording a string section is seen in Figure 5. While this multi-microphone approach may worry some "purist" engineers, it does provide a high degree of flexibility in string recording. Individual microphone inputs are panned to their respective positions in the stereo array. In Figure 5B, the ΔT units represent time delay modules. Those feeding off the stereo mix would be set in the 30-to-50 millisecond range, where their effect would be to simulate early reflections in a concert hall. The delay module in series with the reverberation generator would be set in the 40-to-70 millisecond range, in order to simulate the normal delay in a concert hall before the onset of reverberation.

The string bass may require attention. It is not unusual for studio orchestras to have only a single string bass, and in order to pick up sufficient fundamental sound, a single microphone placed on a low stand in front of the instrument will be needed. Its output will have to be panned and balanced into the stereo array carefully.

For jazz recording, the string bass takes on an entirely different role, and it needs to be highlighted as a virtuoso solo instrument. It is almost always played pizzicato, or plucked, in jazz work, and this demands that finger noises and rattling of the strings on the finger board be picked up accurately, rather than suppressed. A single microphone at a height of about one meter, aimed at the front sounding board, at a distance of about 0.5 meters, will insure a good pickup. Be sure to thin out the bottom end a bit if the sound seems too muddy. ■

RALPH HODGES

Getting Down to Two Tracks Fast and Other Stories

A veteran engineer revives the live mix... and lives to tell about it.

IN DEFERENCE TO ENGINEER Fred Catero, whose story this actually is, I'd like to dedicate this article to all those eager-to-please session people who have been really-and-truly "fixed" in the mix; to all those engineers who have played multi-track musical chairs with a philistine producer; to all those producers who have faced the crack of dawn with the realization that "It's garbage. What I've got here is garbage," and to all those music lovers who bought the album because they heard the group the night before, but now can't discover any similarity whatever.

Who, after all, speaks up for these people? Who offers consolation other than the perfunctory, "It'll be better next time?" No one, really. If that is not the actual reason why the album *Black Pearl* got made, it is at least the message many of its makers hope and trust it will convey.

Black Pearl is an album that no one lost a night's sleep over, because as soon as the band stopped playing, it was done. There were no subsequent mix-downs, no sweetening sessions, no after-the-fact overdubs, and no 4:00 AM erraticisms and coffee nerves. There couldn't be, because the recording was made on two tracks, and what can you do with a two-track tape, except keep it or throw it out? All of which seems to have suited the people involved with *Black Pearl* just fine. Everything was nailed down solid, and everybody went home happy.

A LITTLE BACKGROUND

In the beginning, producer (and prominent jazz critic) Conrad Silvert and featured soloist Herbie Hancock undertook to do an LP for Victor Records of Japan with the 19-piece Tokyo Union Orchestra under T. Takahashi. In short order, Richie Cole and Slide Hampton were also invited in, along with their arrangements. The recording sessions, to take place at San Francisco's Automatt, were scheduled to correspond with a visit by the orchestra to the U.S.

It was Japan Victor's plan to turn the recording into something of a technical tour-de-force, in order to increase its commercial chances. Word came that they intended to send a PCM recorder along with the orchestra—and, presumably, to stamp "digital" on the record jacket, to arouse a little excitement. Neither Silvert nor recording engineer Fred Catero could see anything wrong with that, and there things stood—until Victor announced that they would not send the recorder after all, but would oversee arrangements to rent one in the States. That was fine too. A short time later they sent notice that the rental idea was out, that the Automatt should proceed to record the tape conventionally, and that perhaps it would be converted to digital back in Japan. Something about this scheme seemed to border on the illogical, to say the least. Consultations were held to find some other technical hook to hang the project on. The U.S. team came up with the notion of doing it two track, start to finish; live mix, in other words. How was that for revolutionary?

Well, at first it was a little too revolutionary for Japan. Could such a thing be done? Catero was cooly confident. He pointed out that there was a time when virtually all records were made that way; that he had in fact participated in that time and could

Ralph Hodges is a freelance writer/editor living in the San Francisco bay area.

Figure 1. Engineer Fred Catero prepares microphone placement before recording the 19-piece Tokyo Union Orchestra at The Automatt in a "live" session that was mixed and recorded direct to a 2-track Dolbyized tape recorder. Recorded in this fashion to capture the interaction and spontaneity of the musicians, the LP features Herbie Hancock, Richie Cole and Slide Hampton with the orchestra performing songs and arrangements by Hancock and Hampton.

still hold a job. Japan was still dubious. The inscrutable Catero was reassuring; of course, there would be no problem in backing up the session with a twenty-four tracker running simultaneously. That clinched it.

As everyone must know by now, jazz has a large and wildly enthusiastic following in Japan. Without waxing overly sociological, I would suggest that it is a valued counterbalance to a certain devotion to formal orderliness that characterizes Japanese lives and affairs. Comparatively, jazz is spontaneous, unstructured, let-it-all-hang-out—a chance to relax and, every now and again, to get riotously and even sloppily drunk. So naturally, when you invite a Japanese jazz orchestra to come play in your studio, its members turn up scrubbed, sober, utterly serious, studious and studied, and on time. Catero couldn't believe it when, at ten minutes to the session, he glanced out the control-room window to see everyone in place and tuning up, with the obvious intention of starting on the dot. He thought his watch had stopped and that he had lost at least a half hour somewhere along the way. This was not the way professional session musicians are supposed to behave.

It was a foretaste of things to come. Did the orchestra know the charts? It did. Was it prepared to play, and play well, even for whole minutes at a stretch? Of course. Was it intending to tie things up in union regulations and grave deliberations over coffee breaks and who pays for the new saxophone reeds? Never entered their minds.

When I spoke with Catero some weeks after the session, he was still high from the experience. It had been three and a half days of near-hitchless wonder and mutual congratulations, and at the end, it was the two-track tape that went gaily forth into the world. Nobody ever mentioned the twenty-four-track backup, and Catero hopes nobody ever will. From here on, it is his story, so I'll let him tell it in something like his own words, complete with outbreaks of spleen, finger-wagging and control-room philosophy. Wish you had been there to hear it in person.

THE MECHANICS OF THE SESSION

I like to use (this is Catero speaking) RCA ribbons on trumpets when I'm close miking, because of the mellow quality they impart to really hot brass. When condensers are worked close, such saw-tooth stuff comes out that the sound almost seems to splatter at times. However, condensers are fine for trombones and saxes: Neumann 87s, and I also like the Sony C22s, especially on cymbals and drums. The Shure SM-56 is a great workhorse that I use quite a bit on drums and electric guitars, principally because of its tremendous headroom and the sort of soft self-limiting action characteristics of its diaphragm construction. But, in general, I don't play favorites. If I need frequency response, almost any good condenser will make me happy. For the rest, a handful of dynamics—and not even expensive ones; $150 or so is okay—should do the job.

For *Black Pearl*, we miked the orchestra in pairs, if not in sections. I prefer this for the sake of the better blend. The only people who got individual mikes were those being featured, or who had to be ridden up and down, or who needed some delay and reverb. As a rule, I try to pick up the electronic instruments directly. With jazz, the dynamic swings can have you down in the grass at one moment and right through the roof the next. A

Figure 2. In The Automatt's Studio A control room are (left to right) Tokyo Union Orchestra leader Tatsuya Takahashi, session producer Conrad Silvert, Herbie Hancock, and Fred Catero, along with members of the Tokyo Union Orchestra.

direct feed makes life a lot easier under these conditions, even if the amp doesn't hum and the speaker doesn't break up. I never have any trouble getting a guitar or bass to sound the way I want through processing on the spot or after. If need be, I'll lay down the track via a direct feed during the session, and then play the track back through speakers later. But for this session, I broke my own rule (which is what rules are for, if you can see a better, more logical way) and miked the guitars. It just seemed to provide a more live, more realistic sound in the context of what we were doing. But the bass and the Fender Rhodes were direct. The Rhodes is a case in point. It has stereo outputs, and with a direct feed the separation tends to be much more distinct than with an acoustic pickup.

Hancock's piano was miked very close; mikes on the high and

Figure 3. Herbie Hancock, who wrote and arranged the material and is a feature performer on Side A of the album, conducts the orchestra.

low strings, and a third underneath, with its phase reversed. I wanted to be able to keep everything the man did, so I went all-out for minimum leakage from the rest of the band. In general, I *wanted* leakage—depended on it, in fact. Studio A is a large space (1,500 square feet) of the sort I tend to call an old-fashioned room: hardwood floors, hard walls with absorption here and there, and so on. Music sounds good in it, and I can *use* this fact, make it a real contribution, just as I can use some inter-mike leakage to create a blend and a sense of spatial continuity. Hence everything was left out in the open: no gobos, no screens of any kind, no isolated drum room. The soloists worked the middle of the room, with the orchestra facing them along the front and rear walls. Here, we did have a bit of a problem with the horns. They were up front, facing the drums way in the back, and they couldn't hear them too well, nor could they deal with the slight but significant time delay. As sharp as they were, they couldn't anticipate the beat in a way crucial to some of the syncopated stuff. Finally, I set up a speaker beside them and fed it just the snare and the kick, and that was enough.

SIGNAL PROCESSING

I use two processors faithfully and usually only two: a digital delay and something for reverberation—a plate or a chamber of some kind. You need both—that initial single delay representing the first reflection and then the drawn-out reverberant tail—if you're going to locate an instrument in a mix and give it a "somewhere," which is what I aim to do. When and how much I process depends on where the instrument is supposed to be. If it's up front, it shouldn't need anything. If I want it in a corner in back, it gets an early initial delay and quite a bit of reverb. If in mid-room, maybe just a touch of reverb and a later initial delay, because there should be no reflecting surfaces nearby.

Earlier, we were talking about the sense of depth in a recording, which is something I completely believe in. First reflection and subsequent reverb are key cues for this sense. Of course, there are others: general character of the reverb, spectral

Figure 4. Tatsuya Takahashi of the Tokyo Union Orchestra.

No way could the sound of *Black Pearl* have been gotten in the semi-anechoic sound-sink that passes for a studio today. Those places eat musical sound, devour its character, and few people seem to understand how much needs to be put back in to make it whole again. The engineer adds a little echo and says that's it. But there's really much more.

Naturally, I have used—and use—stereo and binaural miking all the time; it's uniquely valuable for the realistic depiction of an experience, like a night in a jazz club or at the opera. However, for the ideal representation of music in a plausible space for a commercial recording, I find it completely satisfactory only when I can locate one or two places where I can stand, in relation to the performance, and hear everything exactly as I want to hear it. This is rare. With multi-miking you're not so constrained, and you have a chance to patch up those little anomalies you will hear even in the presence of live music. I definitely do not favor the wholesale glamourization of music or the deliberate distorting of a musical experience—unless the composer's intent is surrealism, of course. But I appreciate the opportunity to make slight improvements here and there, and to eliminate the niggling things that really needn't be there.

Finally, multi-miking is *expected* these days. No longer is a producer willing to interrupt a good take to move the trumpet closer to the mike, and perhaps away from the spot where he can hear the other players he depends on for cues. For better or worse, it's now solely the engineer's business to solve these problems of balance, and to create a balance that would perhaps, be impossible in real life, because you just couldn't position the musicians in that particular way and expect them to play properly.

ON THE MIX

So few engineers are capable of doing a live mix these days. Or even if they could do one, they think they can't because they

characteristic of the entire sound, and even—but not always—loudness. But the first two are of paramount importance. Of course, you can achieve them without signal processing per se. If you want to make someone recede, turn down his mike and turn up the ones he's leaking into. It accomplishes the same basic thing.

I love good acoustic chambers. The problem is that many are not good, and sound tubby or hollow. Another problem is that most have a fixed characteristic. I can't alter the initial delay unless I physically walk into the chamber and move the speaker or the mike, and who has time for that? So I depend mostly on the black boxes. Which reminds me: I have a new toy, an EMT with FRAPs on the plate. Their pickup seems more linear to me, and better on the highs, especially when I'm hitting the plate hard. By now, I've had enough experience with it to say I'm "knocked out." (For more on FRAPs, see *The FRAP Point-Source Microphone* in the December, 1979 **db**—Ed.)

Incidentally, *Black Pearl* was not engineered for any remarkble sense of depth. This is not because it was a real-time mix, but because the people didn't want that. They preferred a refined studio with a plausible ambiance, but everything right there, up front, and alive—without going to excess, of course. The drums, for example, are more forward and certainly wider than you'd have perceived them had you been in the studio. This is deliberate; call it a bit of poetic license, if you will. It's a slight enhancement of reality, but it doesn't overthrow credibility.

CONCERNING STEREO

There is no stereo miking to speak of in *Black Pearl;* instead, there is leakage. I'm appalled by all the multi-track mono I hear, with a guy here, here, here, and there, and dead, empty nothingness in between, with no sense of environment, even if it's *supposed* to be a contrived environment. Used properly, leakage is powerful. It keeps the instruments pinpointed, but it also fills the areas in between and makes the space glow and live. Learning how to control leakage is a matter of learning how to use your mikes (which we'll get to in a moment) and the room.

Figure 5. Members of the Tokyo Union Orchestra.

never have, and that's just as fatal. The only mix they know is the one in which they sit around experimenting, fooling with the bass and drums for a while, trying to add a guitar (maybe it works, maybe it doesn't), perhaps starting all over again, listening to the kick alone for a while, ad infinitum. Where does this lead except to becoming a slave not to the music, which would be fine, but to the technology, which isn't designed for a slave and shouldn't have one? And what effect does all this have on the musicians? Could they go on the road with that mess of mistakes they bring to the studio to record? Certainly not; but they've learned it needn't be any better for the recording session because they know that someone is going to sit down and expend hours and agony fixing it. But is this music they're making? I say no, and I say that when you hear it you can tell it's not.

The glory of *Black Pearl* is that I didn't have to spend any time making the musicians look good when they weren't, and so I could get on with my work. And because I could get on with my work, it was done before it had a chance to get in the way of their music. There is nothing special about a live mix, in my opinion. If the band can play the music, and if it sounds at least acceptable in the studio, then that is a natural way of recording it. Of course, the engineer has to focus his full attention on the project at every moment, and he has to have a reasonable feeling for what to do in a tight spot—which can sometimes mean, do nothing and go with it. He cannot let himself get carried away to the point where he doesn't know where he is and what he's supposed to be doing, and he shouldn't let himself get blown off, and out, by 115 dB levels. (Some people don't realize you can't balance at such levels; the ear acts as a natural limiter and everything sounds balanced.) But with the right attitude and a modicum of skills on the part of everyone, a live mix should glide through takes, playbacks, and approvals like a zephyr through a willow grove. And then everyone can go home and get a good night's sleep.

MIKES AND MIKING

I've never been one to get hung up on *the* microphone—*the* mike without which I refuse to do a session, *the* mike that is the only one that can get the snare right. The issue has no meaning for me. A mike's a tool, not a totem or a voodoo doll. It can be used successfully with understanding of its specific nature, and perhaps unsuccessfully without.

Mostly for my own amusement, I went to the last Berkeley Jazz Festival and miked *Santana* from the audience. For mikes I used what was handiest, which happened to be the ones built into a JVC binaural headset. A few days later, one of the group's managers stopped by. I played him the tape and he was stunned. "Why can't my remotes get that sound?", he wanted to know. Probably because the recording team is always thinking about the truck, the gear, the schedule, and all the rest. All I thought about was where to put the mikes.

I see engineers miking amps by putting the mike an inch away from the center of the speaker. This is fine, if you realize what you're going to get, which is the sound of an area of speaker cone about as big as a silver dollar. Meantime, there are things happening 10, 20, even 50 feet out from the speaker that you don't have a chance of getting. Same thing with a tom-tom. You can space a mike an inch away from it and you'll get a "sound," but it won't be the sound of that tom-tom, because there is a lot more to the instrument than a square inch or two of drum head. I don't hesitate to use close touch-up mikes on a drum set when they seem called for, but I start with an overhead pair to get an idea of what the whole thing sounds like. A good trick I learned many years ago: If you want to record something but don't know where to put the mike, block one ear and stick the other where you think you might like to put that mike. You'll know soon enough whether you want that mike there or not.

I am similarly pragmatic about the environments in which I set up microphones. I have miked a guitar in a bathroom for the (to me) excellent reason that everyone loved the way it sounded in there, including myself. It's all a matter of what it takes. Don't ever forget the easy way out.

Ultimately, I think too many people who use microphones do not bother to learn about them and what they do. They know that a mike has X frequency response and Y dynamic range and whatever distortion. But they can invariably find another microphone that has the identical specs and costs $400 less. Why? I've discovered that, like everything else in this business, some mikes stick to specifications very rigidly. As for others, the specs mean that if you're lucky enough to buy a good one, this is what you get.

What I've always done is to take advantage of the things the better microphones can do. The pickup pattern is very important. I've gone up to drummers between takes and moved a mike a tiny bit, perhaps ¼-inch, and they've looked at me like I had gone crazy.

"What are you doing? What did you just come out and do?"

"I just moved it a little bit."

"What's *that* gonna do?"

"Look, this microphone cost $600, and there's a reason. Just one degree off axis gives me better separation between this tom and the other, and I'm *using* that." I know some engineers who figure that as long as the mike is somewhere in the neighborhood of the device it's supposed to pick up, it'll do it, with no consideration of pattern, of dynamic range, of anything. But it's more important to know what a microphone will actually do, than to know that a Neumann is good on this, a Sennheiser is good on that, or a dynamic is good on drums. Yes, there's nothing more important to this business than a mike, but at today's state of technology, and at today's prices, it's hard to find a microphone that won't do the job, *if* you know how to give it a chance. ■

On most multi-track recording sessions, the outputs of at least a few microphones always manage to get combined—either on-session, or later, during the mixdown. The combinations are for the obvious purpose of getting a better balance than would be possible from a single microphone.

What may not be so obvious are some of the polar patterns that may result when the outputs of closely-spaced microphones are combined. With a little planning, a two-mic combination may yield just the right pattern. However, with a little lack of planning, the combination may produce some unexplained results.

To point out just how combining outputs may produce unexpected results, take a look at the well-known—but often misunderstood—M-S microphone system. M-S stands for Middle-Side, in which one uni-directional microphone points straight ahead at the middle of the group while the other microphone is bi-directional; placed as close to the M mic as possible, but with one of its dead sides aimed at the middle of the group. In other words, neither live end of the microphone is pointing in the direction of the music—an arrangement that may not seem too promising to those not familiar with M-S technique.

To understand M-S, it is important to remember that the output polarity from an audio signal reaching the rear lobe of *any* microphone will be *reversed*, as compared to the same signal reaching the microphone's front lobe. On super- and hyper-cardioid studio microphones, this polarity reversal is probably of no practical consequence. However, when the rear lobe reaches the same size as the front lobe—as on a fully bi-directional microphone—then the polarity reversal must definitely be taken into account. In fact, good set-up practice suggests that the rear lobe of a bi-directional mic be kept at a safe distance from the front lobe of uni- (or other bi-) directional microphones, to prevent unwanted cancellation effects.

But what about *wanted* cancellation effects? When a side-oriented, bi-directional pattern is intentionally combined with a middle-oriented uni-, one of two resultant patterns may occur. Assuming the front of the bi- is aimed to the right side, that lobe will reinforce the right-hand side of the M microphone, while the rear lobe will subtract from the left-hand side of the M mic. The result is a uni-directional pickup pattern, oriented about midway between the M and the S: in other words, a cardioid pattern, angled toward the right-hand side of the group.

However, if the polarity of the S microphone is reversed before combining it with the M microphone, the opposite orientation occurs: we now have a cardioid pattern angled at the *left*-hand side of the group.

Of course, with a suitable matrix network, we can achieve left- and right-oriented cardioid outputs at the same time. Why then not simply use two suitably-positioned cardioid microphones in the first place, and never mind the complexities of the M-S matrix?

The M-S technique provides a measure of control that may not be possible with two cardioids. With the M microphone providing a single mono pickup, the output of the S mic may be adjusted to provide just the right amount of stereo imaging. If desired, the M output can also function as a separate mono feed, with no phase cancellation problems, simply because this mono output is not derived from more than one microphone in the first place.

Later on, if a mono mix is required from the left and right M-S derived outputs, there's still no possibility of phase cancellation, since the two S components simply cancel each other out, leaving just the original M output.

Of course, if the M and S outputs are *not* matrixed before recording on tape, then the two-channel tape may be used later on for either mono or stereo playback: mono, by playing back the M track only, and stereo by combining the M and S tracks during playback. The technique can also be useful if the M-S pickup is part of a multi-track recording—say, a string overdub. With the M track panned center, the S output can be adjusted—as before—to provide the right amount of stereo spread to suit the mixdown. Therefore, you're not locked into the stereo image that sounded right during the overdub.

Since the M-S technique gives us two cardioid patterns which are derived from one uni- (M) and one bi- (S) directional microphone, it follows that an M-S pickup can also be derived from two cardioid microphones. With the microphones angled at + and −45 degrees, combining their outputs provides a single forward-oriented (M, that is) cardioid output. Reversing the polarity of one of the microphones before combining them will give a side-oriented (S) bi-directional pattern. For M-S applications, this may be more trouble than its worth. However, it serves as a reminder of what may happen if an unintentional polarity reversal sneaks into any combination of microphone outputs.

For example, two cardioid microphones may be angled over the top of the drum set, with each microphone feeding a separate track. When the microphones (that is, the tracks) are combined later on, an unintentional polarity reversal may demolish the drum sound, especially if anything important happened to be located dead-center. On the other hand, the reversal may be just what's needed to attenuate some leakage from other nearby instruments. In either case, the polarity reversal makes a figure-8 microphone from two cardioids. Just make sure it isn't happening unintentionally, later on.

For the mathematically-inclined, the formulas for various polar patterns are given here:

$A = 1.00 + 0.00 \cos\theta$ omni-directional
$= .5 + .5 \cos\theta$ uni- (cardioid)
$= .37 + .63 \cos\theta$ uni- (super-cardioid)
$= .25 + .75 \cos\theta$ uni- (hyper-cardioid)
$+ 0.0 + 1.00 \cos\theta$ bi- (figure-8)

For example, what's the relative output of a hyper-cardioid microphone, at an angle of 145 degrees?

$$A = .25 + .75\cos(145^\circ)$$
$$= .25 + .75(-.819)$$
$$= .25 - .614$$
$$= -0.364$$

From the five polar pattern formulas given above, note that the output of an omni-directional microphone is—as expected—always the same (1), and therefore, the output level is always 0 dB down. Also, the output of the bi-directional microphone is simply, $\cos\theta$, which is why it is sometimes referred to as a "cosine microphone."

Keep in mind that these formulas describe theoretically-ideal microphones. In real-time life, there's no such thing as an angle at which a microphone is totally dead. About the closest thing to it is the side of a good bi-directional microphone—a side worth remembering when trying to keep leakage at a minimum. Just make sure it doesn't happen by accident! ■

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GREG SILSBY

Sound Reinforcement and Broadcast Audio at the Republican National Convention

Presenting the trials and tribulations of providing audio for the three ring circus known as the political convention.

IT HAS BEEN SAID that political conventions are spaced four years apart so that broadcasters and sound reinforcement personnel can get a chance to catch their breath before the next one begins. Last summer—the four-year cycle completed—it was time for the Republicans to come to Detroit.

With the Republicans came the media: the stars and the super-stars of newscasting. They were there, of course, to see to it that we saw and heard *all* the events and all the speeches. They were also there to interrupt those speeches (which they knew we would not find interesting), in order to inform us what the speaker was saying, or more often, what he was *not* saying.

Along with the media came a virtual city of engineers and technicians, with sufficient equipment to assemble the complete radio and television studios which would be their home-away-from-home throughout the convention.

The responsibility for providing audio to broadcasters, and sound reinforcement in the Joe Louis Arena, was in the hands of two teams. With producer Jack Kelly in charge, CBS provided broadcast audio and video to the other broadcasters. Video control was centered outside, in a trailer, from which Jack exercised control. Because of the magnitude of the production, only a small amount of audio gear was out in the trailer. Most of the equipment was in the control center, a windowless room directly beneath the speaker's podium.

Figure 1. The presidential podium at the Republican National Convention, miked with Electro-Voice RE18s.

Greg Silsby is market development manager, professional markets, at Electro-Voice, Inc., Buchanan, Michigan.

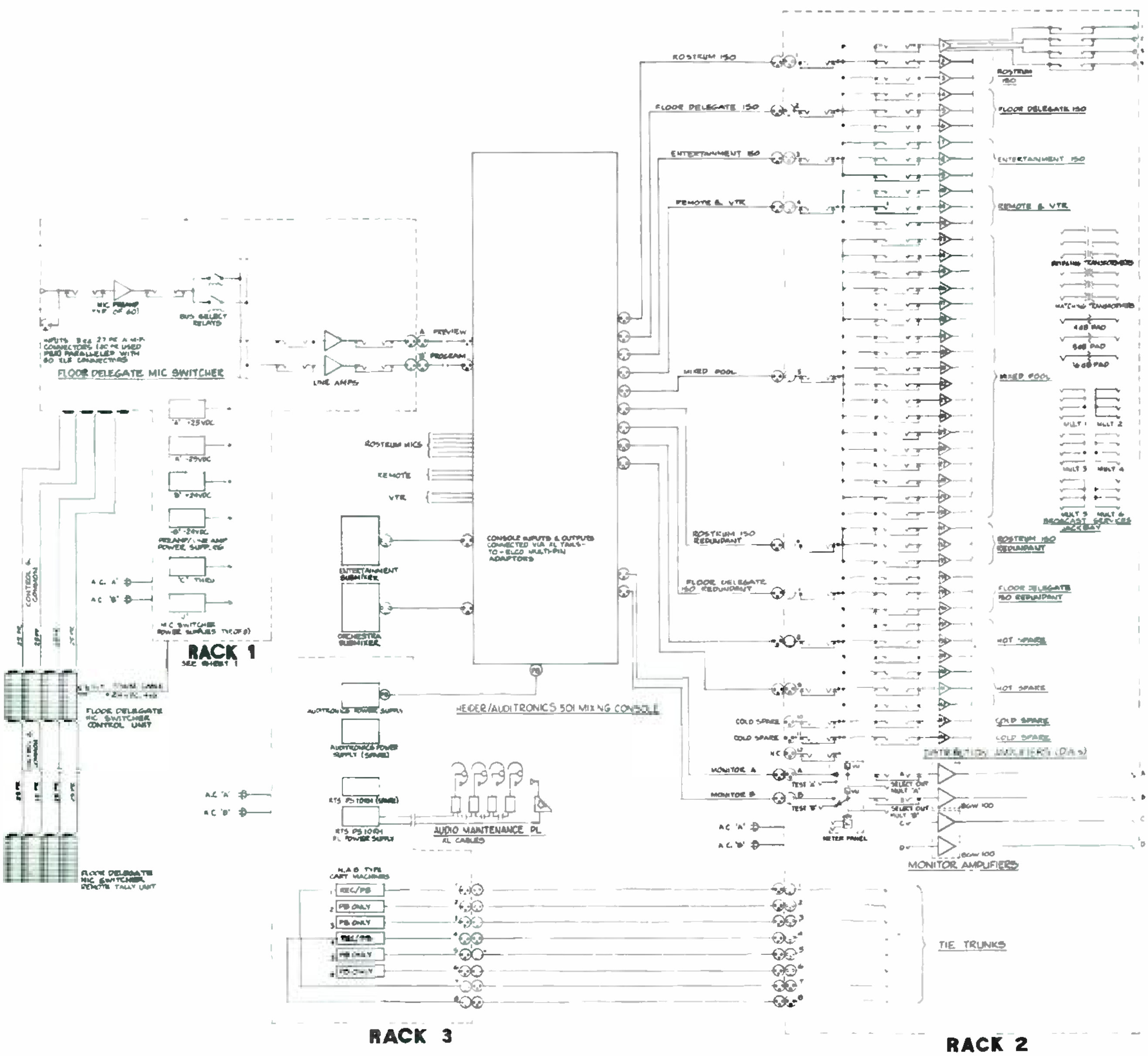


Figure 2. The pool audio system.

Referred to as the "inside pool," this was capably controlled by operations engineer Jim Hargreaves, of CBS Radio and Television, Washington, DC. Jim's crew was made up of CBS technicians from Washington, Chicago, St. Louis and Los Angeles. In addition, he had the assistance of audio services technician Tim Kerr from Filmways, the company from whom the audio-pool equipment had been leased. I was on hand to assist the pool with microphone usage.

Sound reinforcement for the house was provided through the combined efforts of two consulting firms: Coffeen, Anderson and Associates, and KLA Laboratories, Inc., a Dearborn-based contractor.

The technical requirements for broadcast audio were spelled out by CBS and ABC, and then put out for bidding. The equipment package was leased from Filmways, and later on ABC used it to handle the Democratic convention in New York City.

I asked Jim Hargreaves if he had a specific design philosophy regarding the system. "There was complete redundancy. Obviously, each of the components—from a systems standpoint—had to meet broadcast standards, which are somewhat critical, and they did. But along with that was the redundancy, in the event anything failed. The microphone switcher for the floor (you're dealing with 56 delegations) was completely redundant. It was a 60-by-2 switcher. Should anything in either one of the banks fail, the other one was there as a backup. All of the microphone pre-amps were redundant; it was a double-entry system. And in the distribution to the customers—that is, to the broadcasters—there was also the availability of redundant signals.

For miking the delegates, Hargreaves wanted microphones with low handling noise, absence of proximity effects, wide pick-up pattern, and durability. According to Hargreaves, these mics (Electro-Voice DO56 omnis) were the only components in

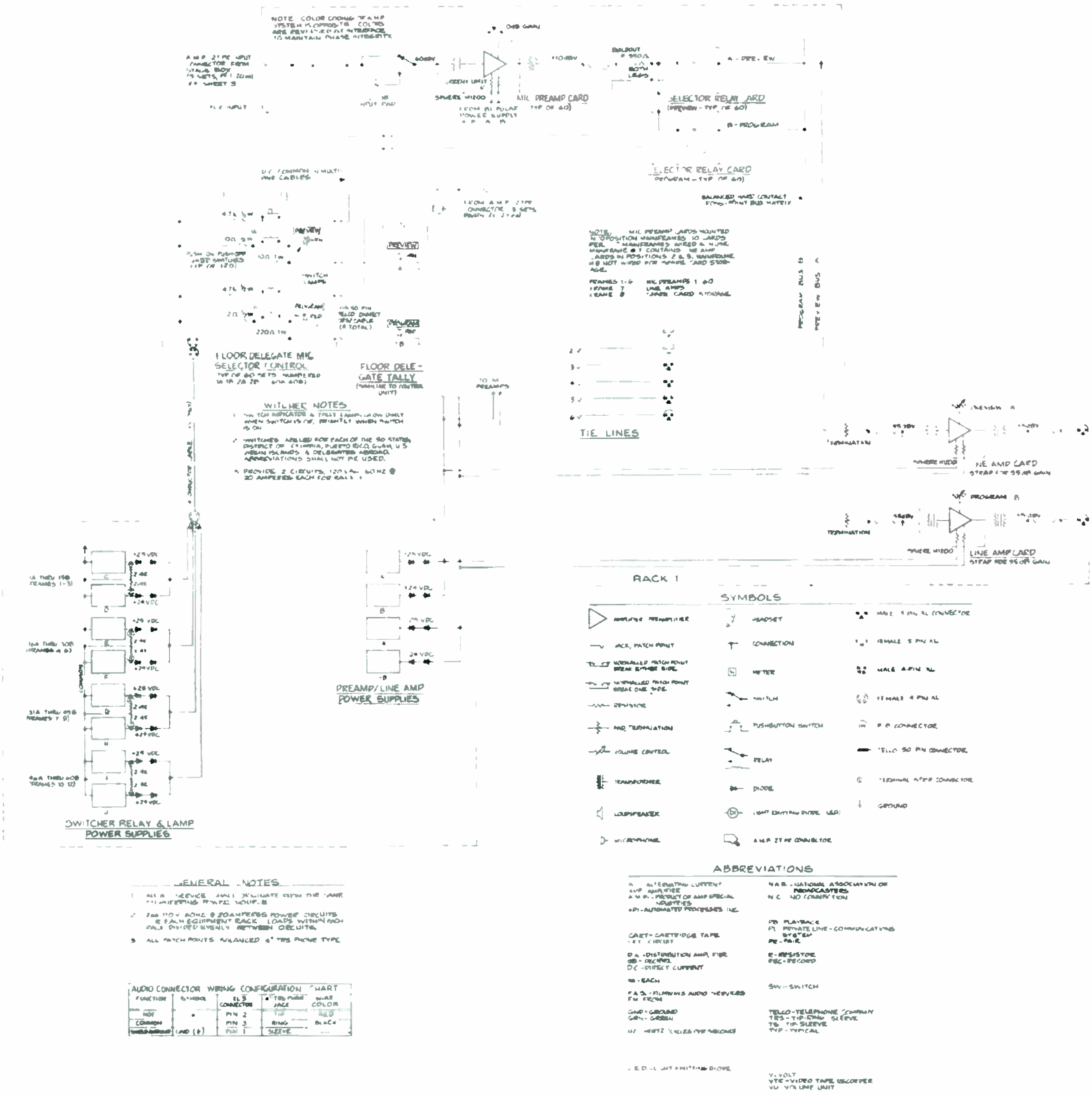


Figure 3. The floor delegate microphone switcher system.

the system which were not doubled. Each microphone was secured to its custom stand by an EV 342 security mount, to avoid easy "rip-offs." They had been modified by removing their on-off switches. The treatment they received confirmed our worst fears, and convinced us we had made the proper selection of both microphone *and* mounting method.

On the podium, there was 100 percent left-and-right redundancy, using RE18 Variable-D® cardioid microphones. An active microphone was mounted on each side of the podium to supply the broadcast feeds. Standby mikes were mounted alongside the actives, to provide a backup system. Fortunately, the alternate mics were never required. An additional RE18 was mounted in front of the podium, to supply the house feed.

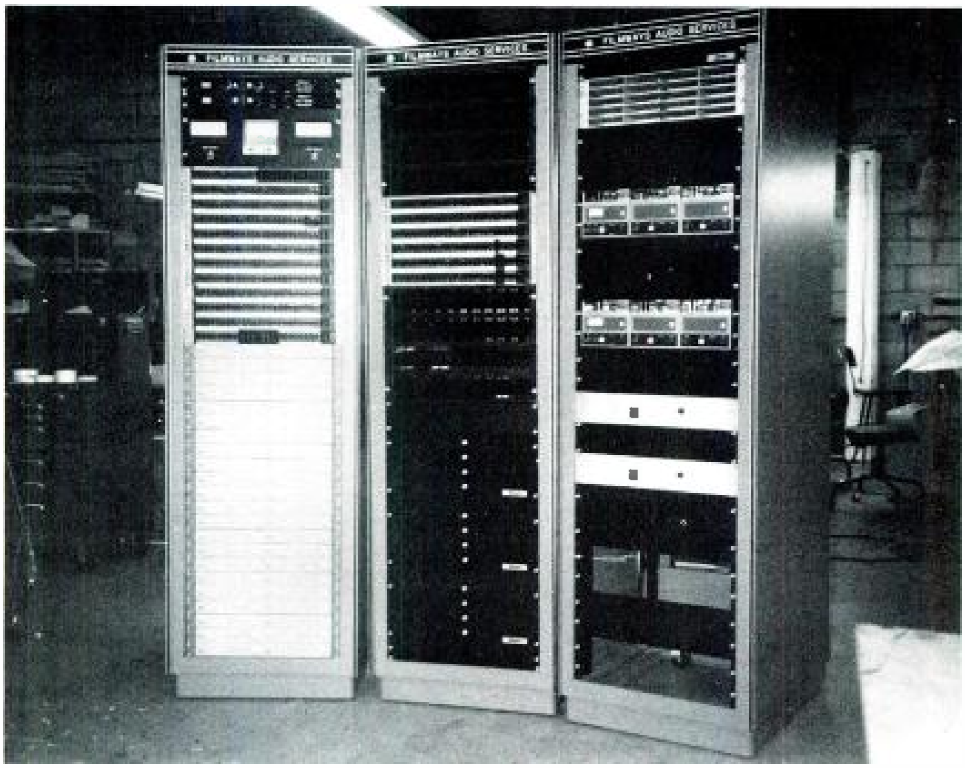
The podium itself had a very "drummy" sound. Because of this, each microphone was mounted in an EV 313A shock mount. The combination of the shock mount and the mic's own internal shock mount resulted in a staggered-resonance system which eliminated all mechanically-induced noise.

The orchestra used a combination of EV RE20, RE10, RE15 and DS35 mics. Guest performers appearing during the convention, including Glen Campbell, Tanya Tucker, Wayne Newton, Donnie and Marie Osmond, and others, used RE18s.

Obviously, no one was going to settle for less than broadcast standards, but shooting for reliability was certainly a key design, installation and operational factor. The audio pool had its own 15 KW standby generator that was checked-out, cable run, then never used.

The sound reinforcement people were fed both output banks of the switcher (redundancy again). They also had a split before the podium-mic pre-amps. If they wanted it, they could choose an orchestra mix or, do their own. Sub-mixes from the audio pool available to the PA also included things like orchestra-minus-entertainers, and entertainers-minus-orchestra.

In addition to microphones, the audio pool had both audio and video tape sound to mix and distribute. Mixing was accomplished on a Heider-modified Auditronics board which



Filmways Audio Services, Inc. supplied the equipment seen here, and Filmways' general manager John Phelan supplied us with the following description:

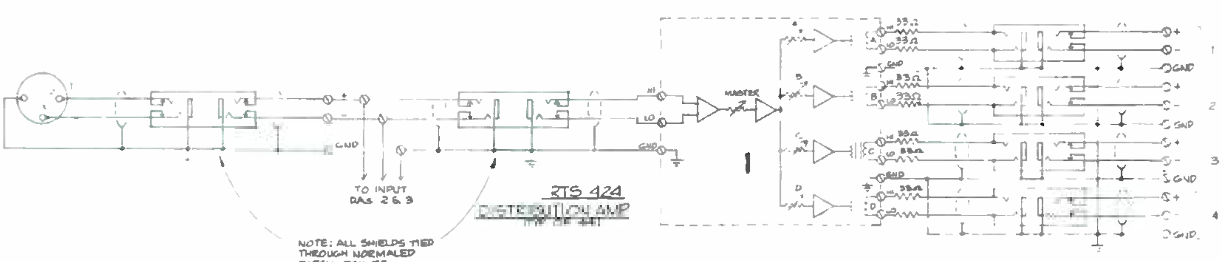
Left-hand rack (Rack 2 of FIGURE 2):
The two top slots are BGW 100 amps for monitoring and checking signal flow. The test panel immediately below the amps has a dbx meter in the center, flanked by two VU meters. Each meter may be switched to either the console monitor (normal position) or to an input jack on the patch bay, to test any of the DA lines. The input of the monitor amps is switched at the same time.

Center rack (Rack 1 of FIGURE 3):
The top portion behind the dark plexiglass is the microphone preamp section. These are Sphere preamps on plug-in cards. Next is the patch bay for this rack, where all preamp ins and outs, and relay ins and outs show up. So do some tie lines and the line amp ins and outs. Below the patch bay are some more preamp cards. All switching relays are located in the back of the rack. The bottom three sections are the Sphere power supplies for the switching relays and the preamps.

Right-hand rack:
This is the utility rack. The top section has storage slots for the NAB cartridges that will be used. Next is a set of three Broadcast Electronics (another Filmways company) tape machines: one record/play and two playback-only. The setup is duplicated in the next section. Next, there are two RTS power supplies for the PL system, and the two empty spaces on the bottom are for the Auditronics console power supplies.

General comments:
All inputs and outputs are through the back in a normal situation. However, the patchbay offers ins and outs as needed. The system was built in 23 days, from design start to the point seen in the photo. The switcher is a 60-by-2 for the delegate mics. The distribution amplifier is 44-by-176, strapped as shown in the drawings. In a modified version, this setup was used for the Baltimore and Cleveland debates. Filmways president Ken Fause designed most of the system to the needs of the ABC and CBS networks.

Below, a detail drawing of the distribution amplifier system, showing full schematic of a single channel. (Photo by Bill Isenberg and line drawings courtesy of Filmways Audio Services, Inc.)



was fed sub-mixes from two smaller mixers, a Neve (for vocals) and a Yamaha (for orchestra). These sub-mixes were also available to the house PA. Delegate microphones were assigned to the main board via the special 60-by-2 switcher with pre-amps. According to Hargreaves, "Each individual switch point was a relay card, so that any given switch point that failed would pop out to allow another card to be plugged in without disturbing the system."

Joe Louis Arena was redecorated with a new speaker cluster just for the convention. I asked Bob Coffeen, "Why did the room need any other sound reinforcement, other than what was already there?"

He answered, "That's the easiest question of all to answer." The main reason was that the loudspeakers in the existing cluster are directed only into the upper fixed-seating areas. "There was absolutely no high-frequency energy directed to the floor, where 2500 delegates sat."

There was a second reason. "The house cluster is in the center of the room and mounted at a height of about 70 feet. Consequently, even if loudspeakers in that cluster had been oriented down toward the floor, the directional realism would have been horrendous; that is, you would have thought everything was coming from over your head while you were looking at the podium."

"Additionally, the echo created by the long delay from that cluster—with sound directed back to the person speaking—would have been very confusing, and would have made it very difficult to speak. It turns out it's easier to speak with a cluster 400 feet away from you than at 80 feet. (That's my opinion.) Once it's 400 feet away, the echo is distinct from what you are saying. It doesn't over-lap very much. And consequently, you can ignore it easier. But when it is about 80 feet away, or something like that, it overlaps with what you are saying enough to confuse you."

The new cluster was located about 52 feet above the floor, or about 42 feet above the podium, and slightly in front of it. The reason for the location was easy: "That's where we could put it. It was alright in that location, but it was a little bit too far into the room. That had to be done to miss lighting and other things. We used the existing cluster to cover the upper fixed-seating areas. We let the new cluster overlap into the lower fixed-seating and let the upper cluster (existing) cover the middle and upper seating."

Coffeen's cluster was made of Electro-Voice components. There were eight TL-606D bass boxes, one TL-606A bass box (pointed down) and a total of twelve high-frequency horns, consisting of 6040As and 9040As with DH1012 drivers. These were powered by SAE P-50 amplifiers, as was the upper cluster, portions of which were delayed with Lexicon time delays to coordinate it with the new cluster. Both Electro-Voice and Crown crossovers were employed. Each cluster was equalized with a White 4001 equalizer plus White 3900 series plug-in units for narrow band equalization.

A ring of small broadcast studios had been set up around the upper perimeter of the arena and these were also fed the house P.A. mix. Again, a Lexicon unit was used to delay the audio which was delivered through existing 8-inch speakers. Each studio opened up onto a balcony overlooking the arena floor, thus requiring the time delay. Each 8-inch speaker also had a control for turning it down or off.

Electro-Voice FM12-3 stage monitors, powered by SAE P-50 power amplifiers, provided some very high monitor levels to both the vocal entertainment and orchestra locations. For some reason, these locations were completely across the arena floor from each other, which meant a lot of monitor or some very upset musicians.

Systems of this size don't just fall into place at the last minute. Eight long days were spent in installing the reinforcement system. This was preceded by two to three weeks investment by KLA in fabricating the cluster and other system parts. Room equalization was done at night. The crew worked three nights, from about four in the afternoon until about seven-thirty in the morning.

Figure 4. Wayne Newton and friend (an EV PL80) during rehearsal for the convention.

The broadcast audio pool started coming together a full sixteen days prior to air time. Hargreaves was there five days before that, to start laying out the on-the-site ground work.

But what the audio crews will long remember may not be so much the long hours and the intense work as the interesting experiences that come out of working a national political convention.

First there was Detroit! The people of Detroit did everything right. They were the perfect hosts and thousands of convention goers will never lose that image of the "Motor City."

For the CBS crew, there was the "Ice Palace," as the cramped control room at times could have been best described. Jim Hargreaves had arrived during a Detroit heat wave to find that the control room had no air conditioning. The temporary system he then had installed was certainly up to the task. In fact, we may have been the only people in Detroit wearing jackets in July. Hargreaves explained, "We discovered we were in the visitors' dressing room in a hocky arena. The visitors, we learned to our dismay, are traditionally given a hard time. They can have heat in the summer, but not air conditioning. All the ducts were there, and the arena was completely air-conditioned, but not in the visitors' dressing room, and that's where we were. So the tradition of hockey caught up to us."

Then there were the bomb "sweeps," when the security officers would follow their sniffing dogs all over the arena. Security, of course, was tight. We had to have three sets of credentials. According to Bob Coffeen, he spent a total of about twelve hours "just trying to get credentials."

We kept on running into different kinds of security guards: Secret Service, plain-clothes Detroit policemen, and who knows what else. Unfortunately, each seemed to respond only to a certain pass, and even with a chain around your neck laden with impressive plastic and paper (plus a Secret Service-issued gold lapel pin and your Junior G-Man secret decoder ring), there were times when you could not get to where you had to be. Hargreaves commented, "We doubled the size of the crew and hoped enough would get through to get on the air."

The fact that both the sound reinforcement and the broadcast audio systems operated so effectively to overcome the many potential obstacles is a credit to the designers, installers and operators. There was the job of reinforcing the voices from the podium, at times over unbelievable ambient noise levels; there were the problems of dealing with inexperienced microphone users; then, there was the RF in the arena. It was almost as if everybody there had a walkie-talkie on their belt or a micro-wave transmitter on their back-pack, but there was never a problem with it.

And where was E. Murphy, with all of those, potential problems surrounding us? I think he just became bored by all the redundancy. He met up with a crew of veterans who saw him coming and wouldn't let him in—or maybe, he didn't have the right pass. ■

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MAJOR NEW YORK CITY recording studio needs additional maintenance men. **Dept. 111, db Magazine, 1120 Old Country Road, Plainview, New York 11803.**

FORMER SERVICE MANAGER of east coast mixing console manufacturer seeks position as service or marketing manager of reputable pro audio manufacturing firm. For resume write to **Dept. 120, db Magazine, 1120 Old Country Road, Plainview, NY 11803.**

WANTED: POSITION as a 1st or 2nd recording engineer, or position with a good professional sound company or professional audio equipment company—installations. Experience in studio recording, live sound reinforcement, video work and audio installations. Good, hard, efficient worker. **Eddie—(205) 263-6353.**

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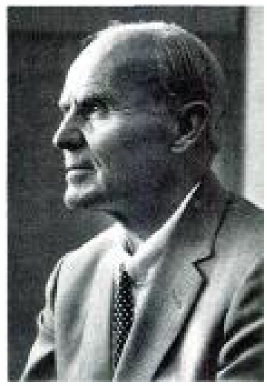
People/Places/Happenings

• **Universal Recording Corporation** has recently purchased a **3M** 32-track digital recorder with 4-track mix down. According to Universal President **Murray Allen**, the digital recorder will be available for use in any of Universal Recording's studios by the end of 1980.

• **Glen E. Meyer**, formerly marketing manager of commercial products at **Electro-Voice Inc.**, has joined the marketing group at **Ivie Electronics** in Orem, Utah. In his new post, Meyer will be responsible for marketing the new Ivie 5000 modular sound reinforcement system. In addition to his marketing responsibilities, he will be working closely with consultants and sound contractors in providing design and applications assistance for the use of Ivie products.

• **Dave Kelsey**, president of **The Filmways Audio Group**, has announced the appointment of **Linda Feldman** to handle marketing for the group. Prior to this position, Ms. Feldman was a communications journalist and marketing consultant. She will be based in the firm's Hollywood, CA offices.

• **Dimension Five Studios**, a professional audio company headquartered in Womelsdorf, PA, recently unveiled plans for greatly expanded coverage and service for Philadelphia's broadcasting, commercial sound and recording studios and announced the appointment of a new general manager and relocation of its Philadelphia offices as major steps towards accomplishing this coverage. **David Meyer**, formerly a northeast regional field sales rep for the **Bose Corporation**'s Professional Products Division, was appointed Philadelphia area general manager for Dimension Five effective immediately. His primary responsibility will be to develop the broadcast and commercial market for sound sales, as well as continuing to service the Philadelphia area professional and musician sound market. Dimension Five will move from its present location in Philadelphia to new, larger offices in Bala Cynwyd.



• **Alexander M. Poniatoff**, whose boyhood fascination with a locomotive eventually led to two major technological breakthroughs as the founder of **Ampex Corporation**, died October 24, at the age of 88.

He founded Ampex in 1944 and served as president until 1955, when he was elected chairman of the board. He resigned as board chairman in 1970 and was named chairman emeritus. Although not active in recent years in the management or administration of Ampex, Poniatoff maintained an office at corporate headquarters in Redwood City, California, participating in several foundations undertaking research in health and preventive medicine.

In 1947, Ampex was down to eight employees in a post-World War II recession, when they introduced the first practical magnetic audio recorder in the United States. That development was followed by the introduction of the first practical videotape recorder in 1956, an invention that revolutionized the television broadcast industry and gave Ampex a worldwide reputation for technical innovation.

Ampex has since grown into close to a ½-billion dollar corporation with worldwide operations and over 12,000 employees. The company's name comes from Poniatoff's initials, together with EX for excellence.

Poniatoff was born in Kazan, Russia, on March 25, 1892. During an interview when he was 84, Poniatoff recalled that he saw his first horseless vehicle, a locomotive, when he was seven. "I decided right then that I would build these locomotives," he told the interviewer.

He attended the University of Kazan, the Imperial College in Moscow, and the Technical College, Karlsruhe, Germany, obtaining degrees in mechanical and electrical engineering.

He was a pilot in the Imperial Russian Navy during World War I, and then in the White Russian Forces that were defeated during the revolution. He escaped to Shanghai, China, in 1920 and worked as an assistant engineer for the Shanghai Power Company until 1927, when he immigrated to the U.S. He became an American citizen in 1932.

He lived in Atherton, California, with his wife Hazel. In addition to his wife, he is survived by a niece, Mrs. Peter (Anna) Kashkadanmoff, of San Francisco.

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